InterestCast: Adaptive Event Dissemination for Interactive Real-Time Applications

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2. Gutachten: Prof. Klara Nahrstedt, Ph.D.

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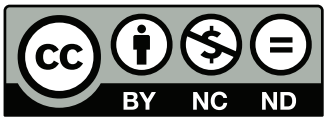

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Abstract

Many networked applications use push-based many-to-many communication. Especially real-time applications, such as online games and other virtual reality applications, need to synchronize state between many participants under strict latency requirements. Those applications typically exchange frequent state updates and therefore require an appropriate dissemination mechanism. Centralized and server-based solutions do not minimize latency, since they always need an extra round-trip to the server. In addition, a server infrastructure constitutes a potential performance bottleneck and thus a scalability limitation. Direct communication between event source and destination is often latency-minimal but quickly exceeds the capacities especially of poorly connected participants because each one needs to communicate individually with many others.

Our proposed solution, *InterestCast*, provides a decentralized event dissemination mechanism that uses peer-to-peer event forwarding, allowing powerful participants to help weak participants with the event multiplication and dissemination. To obtain forwarding configurations that best fit the current situation and application needs, InterestCast adapts them dynamically and continuously during runtime. The application's needs are passed as utility functions, which determine the utility of events with a given latency for a given interest level. Interest levels serve as an abstraction for the importance of events from a specific source, allowing a more fine-grained prioritization than an all-or-nothing subscription model. This is particularly useful if the importance of updates depends on virtual reality distance or another application-specific metric.

InterestCast runs an incremental local optimization algorithm that repeatedly evaluates all possible rerouting operations from the point of view of the respective local node. In each iteration, the best operation is chosen based on the application's utility functions and a system model that predicts the effects of a given operation. As this optimization process is run on each node independently, it scales well with the number of participants. The prediction only uses local knowledge as well as information from the local neighborhood in up to two hops, which is provided by a neighborhood information exchange protocol.

Our evaluation shows that the results of InterestCast's distributed optimization are close to the global optima computed by a integer program solver. Computing the optimum for a given situation globally at runtime, however, is infeasible due to its computational complexity, even with a highly simplified network model. In detailed network simulations, we further demonstrate the superiority of InterestCast over a purely direct event dissemination in online gaming scenarios. In comparison with the direct dissemination, InterestCast significantly reduces the traffic of weak nodes and almost quadruples the possible number of participants for the same average delivery latency of high-interest events.

Zusammenfassung

Viele Netzwerkanwendungen verwenden push-orientierte Kommunikation zwischen vielen Anwendern. Insbesondere Echtzeitanwendungen, wie etwa Onlinespiele und andere Virtual-Reality-Anwendungen bedürfen der Synchronisierung des Anwendungszustands zwischen vielen Teilnehmern unter engen Latenz-Voraussetzungen. Solche Anwendungen tauschen typischerweise regelmäßig Zustands-Updates aus und benötigen einen dafür geeigneten Verteilungsmechanismus. Zentralisierte, serverbasierte Lösungen minimieren Latenz insofern nicht, als sie immer einen separaten Round-Trip zum Server benötigen. Hinzu kommt, dass die Serverinfrastruktur einen potentiellen Performance-Engpass darstellt und damit die Skalierbarkeit einschränkt. Direkte Kommunikation zwischen Quelle und Ziel ist meist Latenz-minimal, führt aber schnell zur Überschreitung der Netzwerkkapazität, insbesondere bei schwach angebundenen Teilnehmern. Denn so muss jeder mit vielen anderen Teilnehmern individuell kommunizieren.

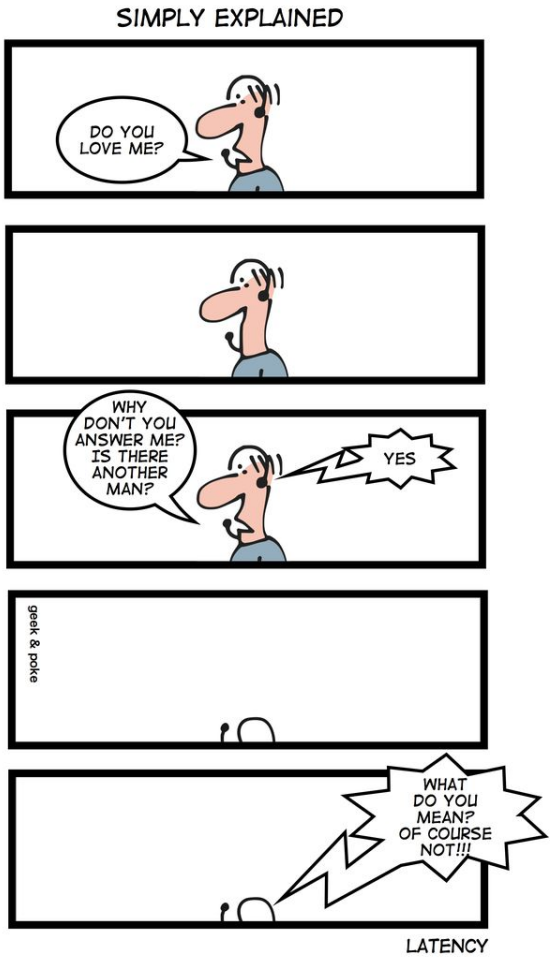
Die in dieser Arbeit vorgestellte Lösung, *InterestCast*, bietet einen dezentralen Mechanismus zur Verteilung von Ereignissen, welcher eine Peer-to-Peer-Weiterleitung von Ereignissen verwendet und so starken Teilnehmern erlaubt, schwachen bei der Vervielfältigung und Verbreitung von Ereignis-Meldungen zu helfen. Um die Weiterleitungskonfiguration zu erhalten, die in der momentanen Situation am besten für die jeweiligen Anforderungen der Anwendung geeignet sind, passt InterestCast diese zur Laufzeit kontinuierlich an. Die Anforderungen der Anwendung werden als Nutzenfunktionen spezifiziert, die den Nutzen von Ereignissen mit einer gegebenen Auslieferungsverzögerung bewertet, abhängig vom Interesse am jeweiligen Ereignis. Interessensniveaus dienen als eine Abstraktion für die Wichtigkeit von Ereignissen von einer bestimmten Quelle und erlauben eine Priorisierung der Auslieferung von Ereignis-Benachrichtigungen. Dies ist besonders nützlich wenn die Wichtigkeit von Ereignissen z.B. vom Abstand der Teilnehmer in der virtuellen Realität oder von sonstigen anwendungsspezifischen Metriken abhängt.

InterestCast führt einen inkrementellen lokalen Optimierungsalgorithmus aus, der wiederholt alle möglichen Veränderungen der Weiterleitungen aus Sicht des jeweils lokalen Knotens auswertet. In jedem Durchlauf wird anhand der Nutzenfunktionen und einem Systemmodell die Operation ausgewählt, von der die größte Verbesserung des Gesamtzustands erwartet wird. Da dieser Optimierungsprozess auf jedem Knoten unabhängig ausgeführt wird, skaliert er gut mit der Teilnehmerzahl. Die Vorhersage verwendet nur lokales Wissen sowie Informationen aus der Nachbarschaft in bis zu zwei Schritten, welche über ein entsprechendes Austauschprotokoll gewonnen werden.

Unsere Auswertungen zeigen, dass die Ergebnisse der verteilten Optimierung von InterestCast nahe an den globalen Optima liegen, welche wir anhand der verwendeten Modelle mit Hilfe eines Solvers für Integer-Programme ermitteln. Die Lösung des globalen Optimums zur Laufzeit ist jedoch unpraktikabel, da die Komplexität des Problems selbst mit einem stark vereinfachten Netzwerkmodell zu äußerst langen Berechnungszeiten führt. In detaillierten Netzwerksimulationen zeigen wir außerdem die Überlegenheit von InterestCast über eine rein direkte Verteilung der

Ereignisse in Online-Spiel-Szenarien. Im Vergleich mit der direkten Verteilung reduziert InterestCast die nötige Datenmenge bei schwachen Knoten erheblich und vervierfacht beinahe die mögliche Teilnehmerzahl bei gleichbleibender durchschnittlicher Auslieferungszeit von Ereignissen von hohem Interesse.

Contents



Simply Explained: Latency.

Oliver Widder, http://geek-and-poke.com/, CC BY 3.0

1 Introduction

> There is an old network saying: Bandwidth problems can be cured with money. Latency problems are harder because the speed of light is fixed—you can't bribe God.
>
> *David Clark*

The major trends of the Internet in the 1990s and 2000s have been on the globalization of data, making services available independently from the consumer's location. This led to the huge success of the Internet. The assumption, however, was that applications could work with moderate network bandwidths and be mostly insensitive to network latencies. With its success, more and more demanding applications became part of the Internet. New web technologies enabled new classes of interactive applications such as Google Docs[1], more and more business logic with increasing real-time demands is run over the Internet (e.g., real-time business intelligence [12]), and applications based on virtual environments, especially large real-time online multiplayer games have become ever more popular [160]. In contrast to most conventional Internet applications like newsgroups, the web, or web services, which use mostly pull-based communication, interactive and real-time applications need *push-based* communication schemes to inform participants about events as they happen. Further, interaction is often *multilateral*, in contrast to applications like one-to-one chats, voice, or video calls. Certain events or actions are relevant for a potentially large group of users, and each user may have a unique set of interests. Online games are in particular prototypical for this set of properties.

From the network perspective, those applications have increasing demands in terms of network bandwidths as well as latencies. Bandwidth increases due to network developments and large scale investment and is, therefore, becoming less of an issue nowadays, at least in fixed networks of well-developed regions [58]. Latencies, however, remain limited by the speed of light. There are ambitious initiatives to bring latencies close to what is possible at the speed of light on linear distance, e.g., using microwave links [145]. Such approaches, however, are associated with high costs and therefore only available to selected services, like high-frequency trading [39]. An alternative approach is to bring the services closer to the consumer. One of the first and largest deployments of such an approach was made by Akamai [126], bringing static content close to the consumer with the goal of a faster, more reliable, and more efficient delivery. Such solutions are application-specific in that the distribution of service functionality must be selected in accordance with application needs. Static content, for instance, can be arbitrarily replicated, while the *direct interaction* between two users is still constrained by the network latency between them.

[1] Google Docs. http://www.google.com/docs/about/

If interacting parties are geographically distant from each other, the possibly significant speed-of-light latency is inevitable. On the other hand, if they are close, geographically or in terms of network distance, there is a huge potential for making their communication more direct. Indeed, many applications, even though they are globally available, have a large ratio of regionalized communication. One example are online social networks, where locality refers to the network of friends. News or posts from close sources in terms of the social graph tend to have more importance and are treated with a higher priority [89]. The algorithms behind online social networks therefore mimic the natural human behavior regarding information valuation. The increasing usage of mobile Internet services also results in a trend to information locality, most explicitly in the form of location-based services [142].

Another example for localized communication are machine-to-machine communication networks, such as wireless sensor networks or vehicular (ad-hoc) networks. Having started with communication technologies and protocols tailored for their specific application needs, these application areas increasingly adopt Internet technologies. Using Internet protocols, all the billions of devices can theoretically talk to each other, which is part of what the 'Internet of Things' [165, 8] and the 'Internet of Everything' [55] stand for. Actual applications, however, often need localized communication. A brightness sensor most likely adapts local lamps, not those on the other side of the planet. Further, machine-to-machine communication often has to deal with high-frequency event streams. Such events may have small payloads, but sending each event individually using Internet protocols may induce a significant communication overhead due to protocol headers.

An application class with similar requirements are *interactive real-time applications*, such as online games, or more generally multiuser virtual environments. To keep the users' views synchronous, the clients regularly exchange update events, often several events per second. Interactive real-time applications further depend on low event delivery latencies. Their scale reaches from a handful of participants up to several ten thousands [53], and the distribution of users ranges from a single room to worldwide.

Most of today's online services rely on server infrastructures acting as mediators between the participating clients. This has the benefits of a central authority and maintenance, as well as lightweight client-side applications. With cloud infrastructures, server management became cheaper and scalable, as virtual servers can be dynamically allocated in large data centers. The cloud economy is based on a small number of such large data centers. Although large operators maintain multiple data centers in different parts of the world, the typical distance to the user is still between a few hundred to several thousand kilometers [107]. Amazon, one of the largest cloud service providers, for instance, has at the time of writing eleven data centers worldwide [5].

While many use cases still require central synchronization, e.g., for data consistency or accounting, interactive applications can still profit from unmanaged high-frequency, low latency updates. Consider a real-time multiplayer game where players navigate and interact in a common virtual world with their avatars. Awareness of the activities of their avatars' surroundings is an important factor for a good gaming experience. Therefore, a high view synchronicity is desirable, which generally requires a high frequency and low latency of position updates among players.

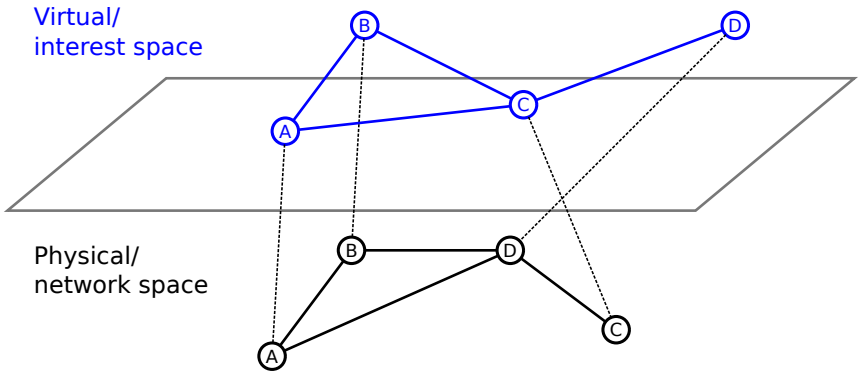


Figure 1.1.: Illustration of the physical space and virtual space in which participants are located. The virtual space is defined by the application and determines the interest of the participants in each other. The physical space is determined by the physical location of the participants' devices and the connecting network. In some cases, the virtual and physical locations correlate (participants A and B), while in other cases, they are completely independent (C and D).

We therefore argue that the dissemination of such time-critical update events should be done using direct communication where suitable. Despite its potential for significantly lowering dissemination latency, aiming for direct communication results in several challenges that must be overcome.

Their composition of properties makes online games a well-suited use case for this thesis.

1.1 Challenges

Our targeted applications have a *push*-based *many-to-many* communication pattern. This means that each participant's updates are of interest to multiple other participants and vice versa that each participant is interested in the updates of multiple others. Considering only network latencies, delivering each event as a direct message from source (i.e., participant of interest) to destination (i.e., interested participant) is the fastest option in most cases. This approach becomes more and more suitable with today's bandwidths. Nevertheless, we have to be able to deal with nodes whose bandwidth is the limiting factor. As soon as the spare bandwidth becomes scarce, *queuing delays* can significantly increase the total delivery latency, and especially weak nodes may not be able to serve all destinations at all. This problem is amplified by the fact that event messages, which are transmitted frequently but often have only small payload data, are inflated by network *protocol header overhead*. As a consequence, the effective net bandwidth in terms of event payload is diminished.

Application-level multicast approaches can mitigate this problem by using stronger nodes as message forwarders and multipliers. By taking network latencies into account, the additional latency introduced by the extra hop(s) can be minimized. Figure 1.1 illustrates the mapping between *physical* (network) and *virtual* (interest) space. While the latter determines who needs to communicate with whom, the former determines the latencies between the participants. Both need to be

taken into account to optimize latencies depending on interest in the virtual world based on the network conditions in the physical world.

Most existing multicast solutions do not consider multi-source deployments. Although a multi-source problem can be modeled as multiple single-source problems, this does not take into account the potential for aggregating multiple messages from different sources to reduce the relative network packet header overhead. Other approaches, such as distributed publish/subscribe broker networks, assume the available additional infrastructure. In addition, there is a delay due to the publish/subscribe mechanism that relies on multicast or in the worst case on sequential unicast. To keep the solution cheap and easily deployable, we strive for a decentralized and self-managing solution.

From the application perspective, on the other hand, there can be *interest gradations* in the sense that update events from a particular source are more important for some receivers than for others. An example are virtual spaces in which the perception of events depends on distance. While actions in the close proximity of a participant are critical, peripheral events may be of less importance. This should be considered by the dissemination solution. Since different applications have different needs in this respect, the application should be able to determine the goal for which the dissemination is optimized. Such option is not available in existing multicast systems, especially in decentralized solutions. On the other hand, there are special-purpose approaches for multiplayer online games and virtual environments that incorporate mechanisms for the differentiation of fidelity levels for neighbors, e.g., depending on their distance and interaction. Those, however, are not generalizable for different applications.

Additionally, the *distribution of load* among participants should be fair. Fairness can be measured in different ways, and existing systems considering fairness usually select some more or less reasonable way for measuring and optimizing fairness. Different applications may have different goals with respect to load distribution. Therefore, the application should also have means for influencing the way the system distributes load. This includes the option of trading the load of the participants for the performance of the system.

Finally, using *best effort networks* like the Internet, the system must cope with varying network conditions. The solution should, therefore, be adaptive to both the network conditions and the application load.

A promising option for tackling adaptiveness with respect to both application requirements and network conditions in a generalizable way is the use of *utility functions*. They allow the application to define high-level utility indicators without the need for dealing with low-level decisions and thereby leaving the system room for self-optimization. Furthermore, they allow the consideration of costs on the underlying system, i.e, the network and node resource usage.

Though being powerful, for an effective application of utility functions, several challenges have to be overcome. The first is to find an appropriate specification level. Applications should only be faced with the parameters that are relevant on the application level. If the system internally deals with lower-level parameters, there should be a translation to application-level metrics. Ideally, there should be no need to provide heuristics in the utility functions. Instead, the system model should be capable of predicting the system response in terms of the utility function well enough

so that the optimization goal can be directly encoded in the utility function(s). Finally, in our case, the system optimization needs to be performed in a distributed setting. In a system with decentralized control, there is no entity with full knowledge about the system state. The utility prediction, therefore, has to work with the limited knowledge available locally.

1.2 Problem Statement

In this thesis, we focus on application use cases requiring *timely many-to-many event dissemination*. Many-to-many refers to the fact that each participant is both an event producer and consumer. Further, each participant has its own individual interest set, making conventional group communication inefficient. Since latency is critical for interactive applications, it is the core optimization objective with respect to application performance. Building on top of Internet infrastructure only allows best-effort guarantees, a fact that is considered throughout this work. Besides performance, it is necessary to consider capabilities and costs of the underlying network infrastructure. We assume bandwidth availability and link utilization as the main factors that determine the costs.

Latency can be minimized using direct connectivity among participants. Hence, this option should be preferred where feasible. Heterogeneous connectivity and device capabilities, however, limit the possible direct fan-out. To mitigate this limitation, participants can help each other by forwarding and multiplying each other's event messages. This distributes load amongst the participants and also yields savings in data traffic due to aggregation.

A dynamic trade-off between the two goals, latency and traffic minimization, is a core objective of this thesis. Such a dynamic trade-off calls for an adaptive solution based on both application demands and network environment capabilities.

Based on the above challenges, we derive the following set of research questions to be addressed in this thesis.

- We have identified the need for an efficient many-to-many event dissemination in decentralized settings. Therefore, the top level question addressed here is:

  How can a decentralized many-to-many event dissemination be solved efficiently and with a particular regard to dissemination latencies?

- To evaluate and improve possible solutions, it is necessary to identify the criteria based on which the goodness can be quantified. This is of particular importance for a self-optimizing solution. Hence, the question is:

  What are the goodness criteria to be considered for the targeted class of applications?

- Application-level utility functions promise to be a flexible way for defining application needs for an adaptive system. In the context of the targeted application the following research question is posed:

  How can application demands and adaptivity goals for a many-to-many dissemination infrastructure, such as interest gradations and bandwidth-latency trade-offs, be formulated as utility functions?

- From the engineering perspective, there is the need to define an appropriate application interface, based on the goals and the requirements identified in the application use cases. The corresponding research question is:

  What is a suitable decomposition and interfacing for a many-to-many event dissemination mechanism providing application-defined optimization goals?

- Having the goal of providing a solution that requires no central coordination, the utility-based optimization has to be able to run in a decentralized setting:

  Can the optimization of the dissemination topology be effectively performed with a local optimization algorithm, i.e., by evaluating the utility function on each node based on its local knowledge?

- Given the targeted application scenarios, there are opportunities to leverage properties specific to those applications. Most importantly:

  How can the clustering of interest be exploited to reduce bandwidth usage with little impact on latency?

1.3 Approach

We introduce *InterestCast*, a decentralized opportunistic many-to-many message dissemination middleware, following the introduced concepts. InterestCast introduces the concept of *interest levels*, which determine the importance of update events of one participant to another. Setting interest levels is comparable to subscriptions in publish/subscribe terminology, but adds the notion of priorities. This concept further provides an interface for decomposing existing overlays for networked virtual environments.

InterestCast adapts to both the application load and the underlying network properties (Figure 1.2). Further, it allows the application to specify *utility functions*, giving the application control over the desired trade-offs. While the first two factors are determined at runtime, the latter provides variability at design time. The utility functions consider event dissemination metrics, most importantly their end-to-end latency, as well as the nodes' load in terms of link utilization. The application specifying the utility functions can benefit in several ways: varying application requirements can be expressed as utility function changes, the utility functions have an influence on the load and performance distribution among nodes and therefore the overall fairness, and they allow tuning the trade-off between performance (low latencies) and cost (bandwidth usage).

For the adaptation, InterestCast runs a continuous and incremental *local optimization algorithm* of message forwarding rules between nodes. All participating nodes repeatedly evaluate and rank possible optimization operations based on their local views and using the application-defined utility functions. In each iteration, the option with the highest expected change in utility is executed, provided the change in utility is positive.

Finally, to minimize message overhead and therefore maximize the effective bandwidth while keeping delays low, a message *scheduling* and *aggregation* mechanism reduces the transmission of redundant data.

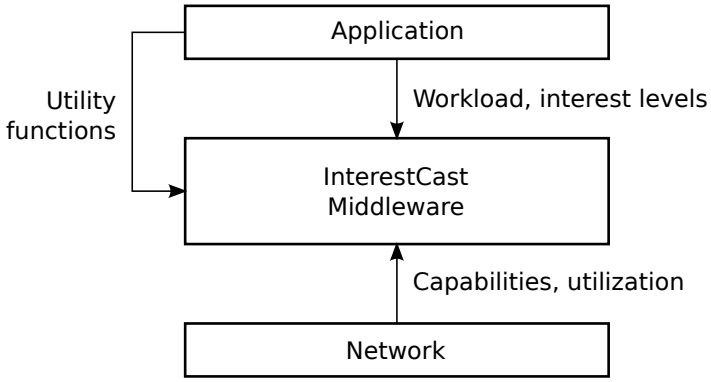


Figure 1.2.: InterestCast's optimization acts upon three main factors: application-defined utility functions, the application workload, and the network infrastructure capabilities and utilization.

1.4 Methodology and Thesis Outline

To set the detailed goals of this thesis, we must first identify the possible target applications. As introduced, the main target application group in this thesis are real-time online games due to their relatively clear set of requirements and conditions. We do, however, consider further application groups: the special case of mobile and augmented reality games, robotics and vehicular networks, as well as air traffic control. Those applications and their particularities are analyzed in Chapter 2. Based on this, the set of requirements is identified.

In Chapter 3, the state of the art of the related work is discussed. Starting with a general introduction to interfacing approaches for event dissemination, publish/subscribe and different multicast incarnations are compared. We further look into specific approaches to adaptive overlays, delay optimization, and peer-to-peer gaming overlays.

Based on the identified requirements, Chapter 4 derives the main design principles for the solution proposed in this thesis. They include basic design decisions such as component decompositions and adaptability approaches, as well as more detailed discussions on topology, routing, and their optimization.

Subsequently, we define the formal system model in Chapter 5. This model provides the necessary abstraction for analyzing and predicting current and future system state. The chapter furthermore carves out the underlying optimization problem. The global optimization problem is formulated as an integer programming problem to be solvable from a global point of view using a standard solver. This allows a comparison between the distributed solutions using local knowledge and the global optimum.

InterestCast incremental optimization algorithm is described in Chapter 6. This includes the basic local optimization algorithm as well as the local utility estimation. The chapter further elaborates on the various options and considerations for selecting appropriate utility functions and presents possible utility functions.

To be able to evaluate and fine-tune InterestCast under realistic network conditions, a prototype is developed, which is described in Chapter 7. The prototype contains the full software stack including connection management, routing, monitoring, optimization, and the application-level interface. The prototype implementation shows the effective cost of the necessary local knowledge, effects of real networking, overhead, and aggregation, and can be used as a basis for real applications.

The prototype implementation is integrated into the Planet PI4 evaluation platform, which is described in Chapter 8. The platform's core is the game Planet PI4, which serves as a workload generator. Alternatively, it provides a workload generation using abstracted mobility models. It allows the execution of the prototype on different network simulators as well as on a real network. It further provides several testing and evaluation facilities like logging, tracing, and analysis tools.

Chapter 9 presents and discusses the evaluation results. We start with a graph-based implementation based on an abstracted network model, showing the basic behavior and the comparison with the global solutions. Subsequently, a more detailed analysis is performed based on the prototype. Finally, Chapter 10 discusses the overall results and concludes this thesis.

2 Background: Applications and Use Cases

This chapter introduces the main target applications and use cases that set the goals of this thesis. We discuss their relevant properties and implications for this work. The set of applications shown here is not intended to be exhaustive, but exemplifies the need for low latency many-to-many communication.

We begin with and put most emphasis on the application domain of online gaming, since this is the primary scenario throughout this work. Mobile and augmented reality gaming is then introduced as a special case of online gaming. Afterwards we briefly examine the application domains of robotics and vehicular networks as well as air traffic control systems.

The chapter concludes with a list of requirements and challenges derived from the discussed applications, which serves as a basis for the design of InterestCast.

2.1 Online Gaming

Since the emergence of the first computer games in the 1970s, there has been a long and successful story of advancements, which has led to a ubiquity of computer games in our lives, driven by a multi-billion-dollar industry [7]. Although computers enable attractive single-user games, which in fact make a significant part of the games market, multiplayer gaming has been a compelling factor from the beginning. Like in conventional games, their main aspect is the competition between human players.

Multiplayer games started as dedicated arcade machines, such as Atari's popular Pong [87, 121]. Games for two players were usually played using two controls on the same machine with a shared screen. Home computers adopted this concept, as do modern game consoles. With the prevalence of computer networks, however, computer games made use of them by distributing the games across multiple computers, providing each player a dedicated device. Early approaches for desktop PCs used direct serial cables or modem connections between two machines. Later came local networks, e.g., using Ethernet.

The Internet created new opportunities by bringing thousands, later millions, and now billions of potential players together. This is when *online games* grew large. Online games range from single-digit to five-digit numbers of simultaneous players playing in a single virtual universe. Games with small and short-lived sessions often use one participant as the master (or server) node. The selection usually happens explicitly in that the player who opens the game session runs the master node. Alternatively, all nodes replicate and process the whole game state and synchronize in a peer-to-peer fashion.

Large and long-lasting games, such as *massively multiplayer online games* (MMOG), typically use dedicated server infrastructures. MMOG worlds are persistent and continuously active, and players

can join and leave at any time. EVE Online[1] is an MMOG with one of the largest game worlds. Its number of simultaneous online players regularly exceeds 50,000 in a single universe [53].

The broader technical term also used in this context is the *networked virtual environment* (NVE), which includes, in addition to online games, social virtual worlds and military simulations [28]. We use the term *virtual world* as the simulated content of the game or virtual environment. In most games, the perception of the virtual world for an individual participant is limited to her immediately surrounding environment. In the simplest case, this is modeled by a circular or spherical *area of interest* (AOI), limited by the *vision range* (VR); sometimes, these are also referred to as *aura*, *focus*, and *nimbus* [19].

The responsibilities of the game server can be coarsely decomposed in collecting updates from the clients, managing (persistent) game objects, selecting relevant game updates for each player (*interest management*), and delivering game updates to each player (*event dissemination*).[2] The largest part of the network traffic is caused by position updates [35, 34], which in their most basic form contain the current position of the player. To keep the views of all players synchronous, it is necessary to exchange those updates regularly. Typical update frequencies are between 5 and 10 Hz [35, 34, 151].

To achieve a high degree of synchronicity among the views of each player and a high reactivity to user actions, a low dissemination latency of update messages is crucial [44, 43]. While for relatively slow-paced role-playing games, latencies of one second can still be acceptable [61], player performance of fast-paced shooter games starts suffering from latencies above 100ms [18] or even less [134]. Omitting the intermediate server for the events between clients, i.e., passing update messages directly between the clients, therefore, helps reduce end-to-end (i.e., client-to-client) latencies. This is where peer-to-peer approaches come into play.

For this purpose, it is unnecessary to fully replace the game server(s) using peer-to-peer technology. Aspects requiring central control, such as authorization, persistent storage, or conflict resolution can remain server-supported, while opportunistic low-latency updates are disseminated in a peer-to-peer fashion.

In the past one and a half decades, research has resulted in a variety of peer-to-peer approaches for the different aspects. An overview on contributions specific to interest management and event dissemination can be found in Section 3.6 of this work. Yahyavi and Kemme [167] provide a recent survey covering a broader range of peer-to-peer online gaming aspects, which go beyond the scope of this thesis.

Packet Overhead

As indicated above, online games use frequent update messages to keep the game state synchronous among participants. Since these messages only need to contain metadata, e.g., player position or object states, they are typically small. Measurements on network traces of real online games show that typical update packet sizes are between 20 and 60 bytes [150, 34, 35]. The

[1] EVE Online. https://www.eveonline.com/

[2] A similar decomposition has been proposed by Fan et al. specifically for peer-to-peer gaming [56]. They distinguish between interest management, game event dissemination, NPC host allocation, game state persistence, cheating mitigation, and incentive mechanisms.

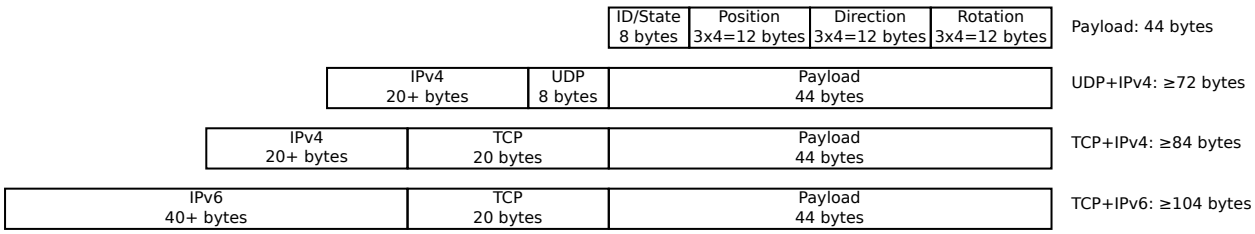


Figure 2.1.: Illustration of network packet sizes for transmitting position updates. Only considering Layers 3 and 4 (network and transport), protocol header overhead is between 50% and more than 100% of the payload size.

average throughput for the activity of a single participant in terms of bytes per second is therefore rather low. 50-byte messages at 10 Hz result in 500 bytes/s or 4 kbit/s net traffic, which has been confirmed by measurements [150, 151, 35].

This traffic profile is distinct from most other common Internet applications. Although the traffic seems low, Internet protocols induce a significant overhead. For example, consider a multiplayer game in which each player's current position has to be regularly disseminated to all interested neighbors. The payload of such update is position data, e.g., a triple of floating point coordinates, plus a player ID and possibly some additional state. To minimize traffic, these high-frequency updates contain as little data as necessary. A typical payload can thus consist of just 20 bytes (e.g., 3×4 bytes coordinates plus 8 bytes player ID). Even a more sophisticated position update packet, as illustrated in Figure 2.1, has a size of just 44 bytes. Sending such an update over the Internet, even using the most lightweight transport protocol UDP, adds another 28 bytes (8 bytes UDP header, 20 bytes IPv4 header). This results in 48-byte and 72-byte packets on the link layer, respectively. Hence, if only a single update event is transmitted in a UDP packet, the size of network protocol headers can exceed the size of the actual payload—and this calculation ignores any protocol below the network layer. Using TCP (20 bytes header) instead of UDP, or IPv6 (40 bytes or more) instead of IPv4 makes the situation even worse: 20 bytes of application payload would be blown up to 80 bytes on the link layer.

Protocol overhead therefore easily adds more than 100% overhead, i.e., more than *doubles* the necessary bandwidth. And this does not even include further overhead of Layer 2 and potential additional encapsulation techniques such as VPN. Prosad et al. [133] exemplify the Layer 2 overhead by showing that the capacity of a 10BaseT Ethernet link is only 7.24 Mb/s for 100-byte packets versus 9.75 Mb/s for 1500-byte packets.

So far, we only considered events from one player. In the client/server case, this is what each client sends to the server. The server returns an aggregated view of all relevant events for the respective client, consisting of larger, but equally frequent packets. Measurements show around ten times higher server-to-client traffic, strongly depending on the game situation [150]. With larger packets, the relative overhead is smaller, which is why server-based approaches work reasonably well with today's Internet infrastructure.

In contrast, using peer-to-peer event dissemination to minimize latencies requires each client to send its updates to each interested peer individually, and vice versa each client receives updates

from many individual peers. In this case, clients have to send each of their updates more than once, and the packet header overheads become highly significant. Miller and Crowcroft conducted a simulation study on this issue [114], in which they conclude that pure peer-to-peer MMOG messaging is not feasible with residential broadband. In their peer-to-peer model, peers subscribe for updates among each other based on a constant-size area of interest (AOI) and deliver updates directly.

Intuitively, the upstream traffic of each peer induced with direct update delivery grows linearly with the fan-out, i.e., the number of subscribers. Inversely, the downstream traffic is linear with the fan-in. Since most residential Internet connections have asymmetric up- and downstream bandwidths, with a significantly lower upstream bandwidth, we will focus on the peers' fan-out. Typical upstream bandwidths of residential connections range from 100 kbps to 10 Mbps. Using the 4 kbps per subscriber from the example above, peers could theoretically serve 25 up to 2,500 subscribers. In practice, however, when saturating the uplink, latencies will skyrocket due to queuing[3], not even considering cross traffic. The simulation by Miller and Crowcroft [114] suggests that already an average bandwidth usage of less than 50% significantly increases transmission latencies. On the other hand, a 10 Mbps connection has a lot of spare capacity in such situations.

In this thesis, the described packet overhead problem is addressed twofold. First, we strive for combining direct message delivery where possible with *aggregation* of update messages by dynamically using forwarder nodes. Those nodes receive updates from potentially multiple sources and can therefore forward aggregated packets. Second, we strive to be adaptive with respect to node heterogeneity. When powerful nodes participate, e.g., one with a 10 Mbps uplink, it is possible to *shift load* from weaker nodes.

Opportunities: Clustering and Interest

With the challenges described above, online gaming scenarios show important properties that we aim to exploit for improvement. The first is the clustering of interest among participants. Since interest is mostly based on virtual world proximity, it is typically highly clustered.[4] This clustering has a direct effect on the number of options for aggregation and load shifting, because forwarding of events can be performed most efficiently using nodes with common interests.

Secondly, the interests of participants in each other typically have gradations. This means that, while some neighbors in the virtual world are of high interest, e.g., because they are close by, others further away, might be visible, but of low interest. Studies from cognitive sciences indicate that the human brain can only focus on a small and constant number of objects simultaneously [157, 138]. The number of neighbors of highest priority is therefore very limited. Updates from the other neighbors are still needed for a reasonable awareness of the environment. Low-interest nodes, however, need lower update frequencies and tolerate higher latencies. Hence, the update dissemination can be optimized based on interest, therefore serving highly interested neighbors with high priority and less interested neighbors on a best-effort basis.

[3] The effect of excessive delays caused by large buffer sizes throughout the Internet has gained attention under the term *bufferbloat* [68].

[4] We have measured clustering coefficients of around 60% in typical scenarios. Refer to Section 9.3.1 for more details.

We have chosen online games as the primary scenario not only because of their popularity, but also because they reflect a unique combination of the different properties and requirements described above. This allows us to study interdependencies between these properties, specifically with respect to latency sensitivity, dynamism, and heterogeneity of both interest and resources. Using the knowledge we have about properties and requirements of specific mobile game types, we can define *benchmarks* that serve as representative and reproducible standard tests [99, 103] to compare networking approaches.

2.2 Mobile Augmented Reality Gaming

A subclass of the above discussed online games are online *augmented reality* (AR) games. Augmented reality games in general augment the real world with virtual objects, which shape the game play. The game therefore is embedded into the real world, giving a new perspective of gaming. This class of games has gained a lot of potential with the ubiquity of powerful enough mobile devices, most notably smartphones.

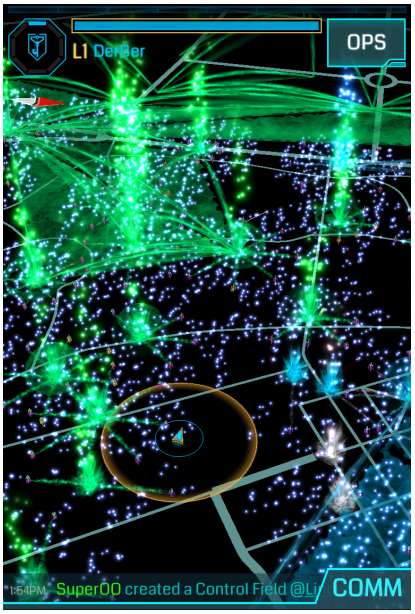
Augmented reality technologies and applications, however, had been on the research agenda already one and a half decades before the emergence of smartphones [11, 10]. Using special-purpose hardware, the use of augmented reality was explored in various application domains beyond entertainment, including manufacturing and maintenance, education, medical, and military. According to Azuma [11], an augmented reality system must have the following three characteristics: (i) combines real and virtual, (ii) is interactive in real time, and (iii) is registered in 3D. Azuma's definition is limited in that it focuses on the computer graphics domain, where the term augmented reality relates to graphical augmentation, e.g., adding rendered 3D objects to live video recorded by a camera. A recent example from computer games that fits this definition is the Android-based game DroidShooting[5].

Putting less emphasis on the graphical representation, an important aspect of augmented reality is the mapping of the virtual space onto the physical space. Large scale, or massively multiplayer, augmented reality games often use the users' GPS-based positions to place them on the virtual map. To ease navigation, the virtual map reflects the real world to a certain degree, usually by using a street map underlay (e.g., Google Maps[6] or OpenStreetMap[7]) for the virtual world map. The most popular example of this class of games is Ingress[8] (Figure 2.2a). TowerWorld [97] (Figure 2.2b) is a prototype of a similar game which was developed for the evaluation of mobile publish/subscribe solutions is this application class.

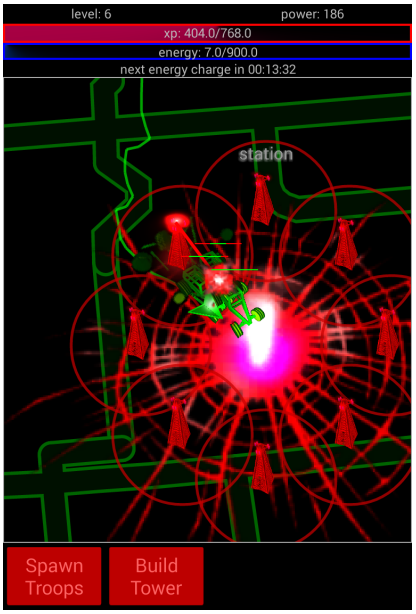
Local Communication

By mapping the virtual world onto the physical world, interest within the virtual world becomes directly related with physical proximity. Interest-based communication (e.g., position updates, player actions) is therefore highly localized with respect to the physical world. Figure 2.3 shows

[5] DroidShooting. https://play.google.com/store/apps/details?id=jp.co.questcom.droidshooting
[6] Google Maps. https://maps.google.com/
[7] OpenStreetMap. http://www.openstreetmap.org/
[8] Ingress. https://www.ingress.com/



(a) Ingress



(b) TowerWorld

Figure 2.2.: Screen shots of two typical mobile augmented reality multiplayer games. Both use the physical location of the player as the in-game position and show the local street map as an underlay of the virtual world.

that a large part of the communication traffic of an augmented reality game has a target of only a few dozens of meters away. This raises the question about keeping traffic local even more than for stationary network games. Yet, today's mobile applications are largely cloud-based, meaning that the network sees only communication between the devices and cloud servers.

The mobile devices' cellular network connectivity is often the limiting—and most expensive—factor for online applications, particularly for interactive ones. Beyond the dependency on a reasonable infrastructure coverage, cellular communication is expensive in terms of energy usage because the connection needs to stay permanently in high power mode despite the low utilization [14, 125]. Furthermore, although low in throughput, the accumulated data volume of long-lasting sessions easily exceeds the volume of inexpensive data plans, jeopardizing the users' desire for playing the game. In a test session, the prototype TowerWorld [97] used around 10 MB per 15 minutes. Users of Ingress, which is less interactive and more optimized than TowerWorld, report varying values of 200 to 600 MB per month for typical usage.[9] Finally, cellular networks often induce higher latencies than wired broadband connections.

Opportunity: Wireless Ad-Hoc Communication

Given those facts, the use of opportunistic ad-hoc networks for a low-latency local communication offload has a great potential [98]. Today's basic technologies that qualify for such connectivity on consumer devices are Bluetooth and WiFi (IEEE 802.11). In the Bluetooth protocol stack, the Bluetooth Network Encapsulation Protocol (BNEP) [23] provides an Ethernet emulation, which

[9] Based on user reports on Reddit. "How much data are you using? How active are you?" http://www.reddit.com/r/Ingress/comments/18u5k7/how_much_data_are_you_using_how_active_are_you/. Accessed 2014-04-09.

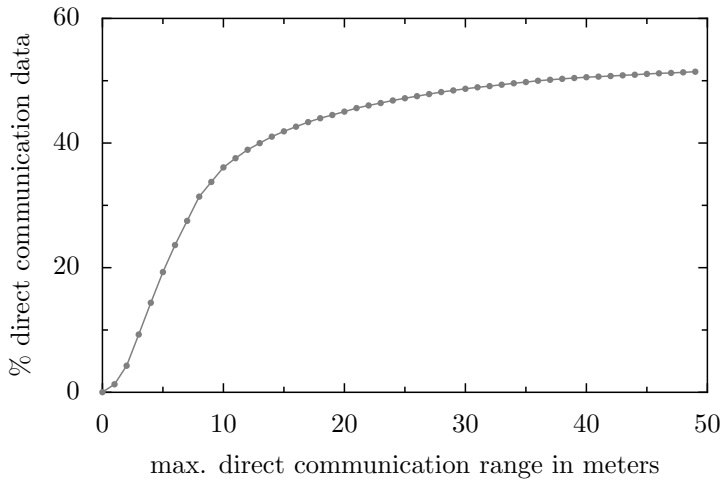


Figure 2.3.: Ratio of data that could have been sent directly depending on the direct communication range, measured in a test session with the augmented reality game prototype TowerWorld [97]. Almost half of the data is sent to a destination less than 20 meters away from the source.

can serve as the base for an IP stack. The need for explicit device pairing, however, limits Bluetooth's suitability for dynamic ad-hoc networks. IEEE 802.11 [79] specifies the Independent Basic Service Set (IBSS) that allows for an ad-hoc communication mode without a coordinating access point. Wi-Fi Direct [166] is an alternative WiFi ad-hoc connection standard that has been promoted by the industry lately. Unlike IBSS, it uses software access points on the participating devices and focuses on easy and secure connection setup. Yet, the dynamically allocated access points, make Wi-Fi Direct less flexible in group size and range. On the other hand, Wi-Fi Direct is available on many recent smartphones and tablets, while IBSS is rarely activated on consumer devices. Hence today, the perfect ad-hoc communication technology that is available on a large number of mobile devices is still lacking. Nevertheless, for this work, we assume the availability of an IEEE-802.11-IBSS-like wireless ad-hoc protocol on all mobile devices.

Wireless multi-hop routing protocols, such as AODV [131] and OLSR [42], enable nodes to communicate with each other even if they are not in direct wireless transmission range. This works as long as there is a chain of intermediate nodes that are in each other's range to forward the messages. Hence, given a sufficient device density, these protocols can extend the effective communication range by orders of magnitude. For augmented reality games, only a small number of hops can be sufficient (Figure 2.4).

Forwarding nodes, however, have to bear additional load. The use of a simple application-level multicast mechanism, e.g., using direct overlay connections from source to destinations, can lead to a multiplication of forwarding traffic. If multiple source-destination pairs use a common forwarder, the forwarder has to carry the same information multiple times. Network-level broadcasts, on the other hand, might be too unbounded in larger ad-hoc networks.

InterestCast can mitigate this problem by aggregating common data on common forwarding links. Provided that forwarder nodes are also participants of the InterestCast overlay, the measurement of overlay link properties (most importantly latency) gives hints about the forwarders in

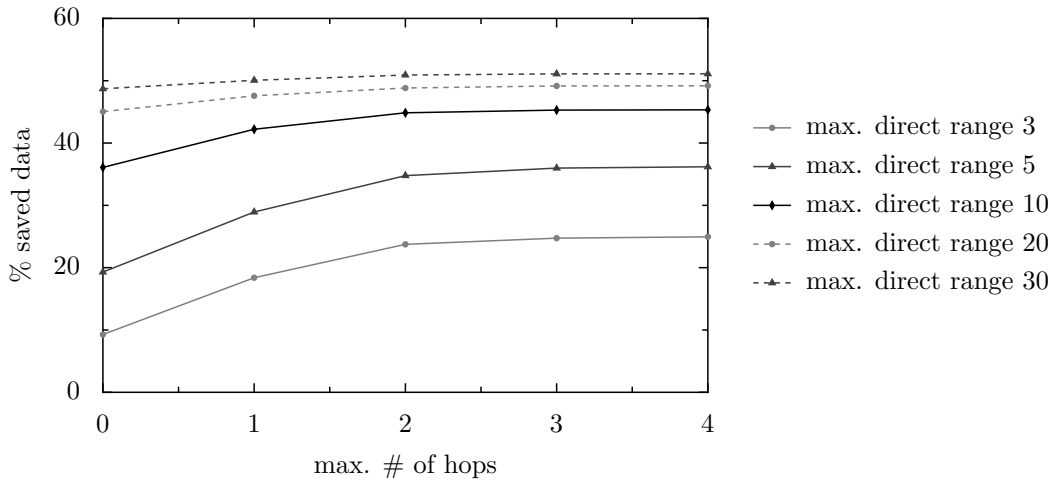


Figure 2.4.: Accumulated ratio of data that could have been sent over a number of hops depending on the single-hop transmission range (meters), measured in a test session with the augmented reality game prototype TowerWorld [97]. A large amount of data requires only a small number of hops.

the underlay. The forwarding function can then be *pulled up* into the overlay, where the necessary information for message aggregation is available. This allows for keeping a general-purpose ad-hoc routing protocol, while at the same time fitting the overlay routing structure to the underlying topology.

Mobile games can benefit even more from localized communication than conventional multi-player games. Although not yet deployed on a large fraction of mobile devices, the basic necessary wireless ad-hoc connectivity and routing technology is available. InterestCast provides an overlay routing mechanism that adapts to the underlying topology.

2.3 Robotics and Vehicular Networks

Multi-Robot Systems

Multi-robot systems [110] have become an important research field in the robotics domain since the early 2000s. Multi-robot systems can be seen as a subclass of multi-agent systems [59], which, as a research discipline, has relations to distributed systems, artificial intelligence, human-computer interaction, and ubiquitous computing. Most practical multi-robot systems considered in research consist of only a small number of robots [17], and most work considering coordination [57] deals with the aspects of interaction languages, semantics, and higher-level coordination tasks.

Basic communication for the low-level real-time coordination, however, is nevertheless needed. Deployments of self-sustaining autonomous robots, e.g., for rescue missions [90], typically use mobile, and possibly ad-hoc communication to be infrastructure-independent. Particularly in rescue missions, the projected number of independent agents is larger (100 or more), control is mostly distributed, and real-time planning is in the order of seconds [90]. Furthermore, robot teams

moving in a real-world space need the concept of positioning and proximity, and coordination is most time-critical where agents are close to each other or form a group. Finally, agent behavior and coordination should be adaptive to their environment. For example, simple situations might be covered with a fast local coordination, while complex situations require tighter interaction and more planning [85]. With the adaptation of planning schemes, communication requirements, such as delay bounds, may also change. With these properties, the concept of InterestCast appears to be a good fit to be used as the low-level distributed status and coordination message exchange mechanism.

Vehicular Networks

Similarly to multi-robot networks, vehicular networks [119, 127] are built among physical, mobile entities—in this case cars and trucks. Due to their participants' mobility, they emerged as a variation of Mobile Ad-hoc Networks (MANETs), named Vehicular Ad-hoc Networks (VANETs). Besides vehicle-to-vehicle (V2V) communication, there is vehicle-to-infrastructure (V2I) communication, which uses cellular or WiFi infrastructure. Among the most prominent applications of vehicular networks are safety and traffic optimization applications such as accident, congestion, and road condition warnings [162], which are propagated locally in real-time. More recently, entertainment applications, including distributed gaming gained increasing interest [127].

The immediately critical but rare communication, e.g., accident alerts, will most likely use lower layer forwarding mechanisms to achieve the best reactivity and reliability without any trade-offs. On the other hand, regular monitoring and control information dissemination, as well as mobile entertainment applications, such as mobile gaming discussed in the previous section, could benefit from a system like InterestCast. Those non-safety-critical applications require trade-offs in the resource usage between applications with different communication requirements, e.g., with respect to bandwidth and latency. Local traffic safety monitoring, for instance, uses periodic messages containing vehicle speed, position, and direction [4]. In addition, event-driven messages are sent once unsafe situations are detected. Especially vehicle data like position and speed is most relevant for other close-by vehicles, while more distant vehicles in the same traffic flow might still be interested, but timeliness is less critical. Furthermore, information from vehicles ahead in traffic is more informative than from following vehicles. Finally, groups of vehicles moving in the same flow have the highest interest in each other's information and therefore form clusters. Those interest distributions are similar to those discussed in the context of virtual game worlds above.

2.4 Air Traffic Control

A timely and scalable event dissemination is a core requirement also for modern aeronautical control systems [70]. Much data is produced and consumed on a spatio-temporal basis, i.e., with respect to location and time. Trajectories can be modeled using 4D coordinates (longitude, latitude, altitude, time) [156]. For example, up-to-date airplane positioning information allows making route planning more dynamic to meet near-term needs. Information about detailed weather conditions, which are scarce across oceans, can be observed in real-time by airplanes and disseminated to following airplanes.

Much of this data might be of little immediate interest to central control authorities and particularly to other air traffic participants that are not in close range to the reference location, supporting the case for localized dissemination. Generally, air traffic control currently aims for a more decentralized coordination, for which a decentralized event dissemination system could be an important part.

2.5 Requirements and Challenges

From the targeted applications and use cases detailed above, we identify the following set of requirements and challenges. They are used as a basis for the selection of potential approaches and design decisions, which are discussed in Chapter 4 of this thesis.

Latency sensitivity. The targeted applications are sensitive to latency. In many cases, this means that the goal is to minimize delivery latency. When disseminating events to multiple receivers, minimizing can refer to the average latency but also the maximum latency among the receivers. Depending on the particular application, there might be fixed upper bounds for event delivery. For particular reasons, such as fairness among participants, it might also be desirable to minimize the latency variance among receivers.

Best-effort adaptation. There are physical bounds on the latency and throughput the underlying network can provide. Since typical network infrastructures, such as the Internet, do not give latency guarantees or predictions, overlays can only provide best-effort guarantees.

The dissemination overlay should therefore (1) be prepared for different—and possibly changing—application needs and (2) adapt to changing conditions of the underlying network to provide the best possible performance in all situations.

Many-to-many communication. Every participant is an event producer and has an individual set of interested participants. This many-to-many communication scheme generally does not lend itself for an efficient clustering into a fixed set of multicast groups or channels. Instead, each participant needs a separate dissemination structure.

Many scenarios, however, show a strong clustering in the participants' interests, induced by the interest locality. This fact can be used to facilitate the maintenance of the individual dissemination structures.

Dynamics. Locations and interests of participants are highly dynamic. In particular, the interest sets can change constantly. The reorganization of dissemination structures should thus generate little overhead.

Rebuilding the dissemination structure from scratch on each change is not an option. Furthermore, a quick fallback mechanism should compensate failing nodes or connections.

Packet overhead. Update events are often small, because they either contain incremental updates or very limited state (such as coordinates of a position). This results in high relative overheads from packet headers of the different network layers. A low ratio between actual payload and the gross traffic unnecessarily reduces achievable node fan-out.

The dissemination system should therefore aggregate packets where possible to maximize efficiency.

Interest gradations. Interest among participants often has gradations. While some (e.g., close players, close friends) are of high priority, delay or message loss from others might be tolerable.

The interface should therefore provide means for specifying the interest priority to be able to tune event delivery to meet the needs of the highest-priority participants first.

Scalability. The total number of nodes in the network can be very large, while only relatively small subsets are of interest for a certain participant. The algorithmic complexity should therefore only depend on the number of interested participants, not on the total system size.

Some participants may be of particularly high interest, causing heterogeneous fan-outs. For instance, in online games, areas with a high player density require many connections between the players. This can easily exhaust the available connection bandwidth of single nodes.

Resource heterogeneity. The participants' resources are heterogeneous. While some nodes, e.g., those on mobile networks, may have extremely constrained connection bandwidths, other nodes have a lot of spare capacity. Ideally, all participants should get the same service quality, independent of their connection and location.

Infrastructure. Dedicated infrastructure (e.g., multicast servers) might or might not be available. The system should, therefore, be able to work independently of dedicated servers. However, if such options are available, it should be possible to leverage them.

3 State of the Art

In this chapter, we discuss and categorize important existing work with respect to event dissemination and the application-specific challenges identified in the previous chapter. We start with the conceptual high-level view, exploring approaches how applications can interface with the dissemination infrastructure. Subsequently, we dive deeper into the lower-level message multicast implementations and further provide an overview of publish/subscribe systems. Finally, we move towards more specialized systems and look into the aspect of interest management and integrated approaches for interest management and event dissemination.

3.1 Interfacing Event Dissemination

Although the concept of delivering events in the form of messages to a multitude of receivers is intuitively comprehensible, a closer investigation of possible options and features poses a multitude of questions regarding design decisions. Early approaches were simple multicast solutions with multicast groups, such as IP multicast [48]. Messages could be sent to a group so that all registered group members receive it. When only unicast is available, the most primitive solution is obviously to send the message to each receiver individually. This requires each sender to know all receivers, but on the other hand allows an individual selection of receiver groups for each message. Such approach, however, limits scalability with respect to the receiver set size. Scalability and implementation aspects are further discussed in Section 3.2. Here, we want to concentrate on what information the application needs to manage and what can be hidden behind the event dissemination facade.

A basic distinction criterion with respect to the interface is the way the receivers for a particular message are selected. The selection can be sender-initiated (i.e., the sender has the information who shall receive the message), receiver-initiated (receivers join a certain sender, channel, or group), or implicit and based on the content of a message. Simple unicast-based dissemination belongs to the first category; group-based multicast solutions (e.g., IP multicast) belong to the second. The third category is largely covered by a variety of publish/subscribe systems.

3.2 Multicast

When it comes to the simple but efficient dissemination of data to a set of receivers, multicast protocols are the means of choice. In contrast to the unselective broadcast (e.g., [65]), multicast uses an explicit selection of the set of desired receivers. By running multiple protocol instances, in principle any number of multicast groups can be created. This allows for a basic channel-based publish/subscribe implementation with little effort, but omits some decoupling of the participants, especially with respect to time (cf. Section 3.4), because messages are not stored for offline participants.

There is a plethora of protocols, especially in the area of application layer overlays, aiming to solve multicast efficiently. Many of them have been analyzed for different application domains [168, 96, 66]. This section therefore provides only a brief overview, focusing on the issues relevant for this work.

3.2.1 IP Multicast

IP multicast [48] is an extension to the Internet Protocol that allows sending IP datagrams to multicast groups. Datagram delivery is defined as best-effort, analogous to IP unicast. Membership in multicast groups is dynamic in that hosts can join and leave groups at any time. Multicast groups (named host groups) are identified with IP addresses from a specific range (224.0.0.0 to 239.255.255.255). Sending a datagram to a multicast address delivers the datagram to all hosts is the corresponding group. To do so, the sender does not even need to be member of the group. This way, IP multicast can work as both single-source and multi-source multicast.

Host groups are not limited in size, but, since they are addressed using IP addresses, their number is limited. The most important practical limitation, however, is that IP multicast has never been widely deployed on the Internet. Without such, it is of little use for large-scale Internet applications. The main reasons are considered to be limitations in access control, security, address allocation, and network management [51]. It is also argued that the end-to-end principle [140], stating that application-specific functions should be implemented on the end hosts rather than in the network, prevails over the efficiency benefit of IP multicast [41].

3.2.2 Application-Layer Multicast

Due to the limitations of IP multicast, a lot of research focus has been put on application-layer multicast (ALM), on which we concentrate in the following. ALM only employs the end systems for data multiplication, therefore it is sometimes called end system multicast. An advantage of ALM over IP multicast is its greater flexibility. While IP multicast has to concentrate on core features to keep in-network complexity low, ALM can incorporate more sophisticated, and therefore more complex, approaches. These include application-specific extensions and optimizations, which explains the large number of variations. In the following, we first give an overview of general-purpose application-layer multicast solutions, before looking more specifically at topology optimization and delay minimization.

Overcast [83] is an early approach for replacing IP multicast. It builds overlay trees, focusing on organizing nodes to build high-bandwidth channels. Fast joining of nodes is an explicit goal, but latency is explicitly not an optimization target. Joining nodes start at the root of the distribution tree and try to move downwards towards the leaves as long as they do not observe bandwidth degradation. This leads to particularly deep trees and therefore high maximum latencies.

The Banana Tree Protocol (BTP) [77] provides a multicast service with best-effort delivery, on top of which a group communication as well as a file sharing protocol are defined. Nodes can switch their positions in the tree to optimize the metrics latency, degree, and total tree cost. The optimization process is performed in a distributed fashion by switching siblings. The authors,

however, do not explain based on which information the selection of options is performed. Although the basic algorithm takes latency into account, the further analysis is only done with cost minimization in mind.

ALMI [130] targets small multicast groups and employs a central session controller for node registration and building the spanning tree. The session controller communicates directly with all members using unicast messages to propagate tree updates. Each pair of nodes regularly sends pings to each other to measure the link delay. To reduce the $O(n^2)$ measurement cost necessary for a full mesh, nodes use a fixed degree and regularly replace known bad edges to converge to the optimum mesh.

Chu et al. present Narada [41, 40], an End System Multicast protocol, explicitly designed as an IP multicast replacement. Narada maintains an overlay mesh among the participating nodes on top of which multicast spanning trees are constructed. The mesh is continuously updated to optimize for latency by adding links where latency is significantly reduced and by removing links that are least utilized. Spanning trees are constructed using a distance vector protocol similar to DVMRP [49]. The authors suggest application-specific customizations with respect to bandwidth and latency prioritization, but only really consider one scheme. Available bandwidth is discretized into a few classes. Links are first selected based on the bandwidth class, and if there are several options left, latency is minimized. Hence, the authors explicitly address the latency vs. bandwidth issue, but always prioritize bandwidth over latency.

NICE [15] targets low-bandwidth applications with large receiver sets and focuses on constraining control overhead, node degree, and latency stretch[1]. Nodes are organized into a hierarchy of layers, in which they are grouped in clusters. Clusters are based on inter-node latencies to minimize forwarding latency stretch. NICE differentiates between the distribution topology, which must be loop-free, and the control topology, which is denser. Nodes join by connecting to the highest layer in the hierarchy and subsequently finding close-by neighbors in one of the lower-layers clusters. Clusters have a minimum and a maximum size. To stay within these bounds, they are merged and split respectively. Although considering latency for the optimization, average path lengths beyond 20 hops with 128 nodes are hardly suitable for latency-critical applications.

OMNI [16] is designed for real-time applications, such as media streaming. It tries to minimize latency with bounded out-degrees, but it assumes dedicated Mulicast Service Nodes (MSN) deployed by service providers. The authors first present linear programming solutions to the NP-hard minimum average-latency and minimum maximum-latency degree-bounded spanning tree problems. The spanning tree only refers to the MSN topology. They then present a decentralized heuristic algorithm to build a spanning tree among the MSNs that converges to the optimum. The algorithm is adaptive with respect to the network latencies and the number of clients at each MSN, but the MSNs themselves are assumed to be mostly static and stable.

SplitStream [32] is a multicast system building on top of Scribe [33]. It builds a forest of multiple trees, each transporting a stripe of the data, so that nodes can decode the data even if a fraction of the stripes is missing. Furthermore, all nodes are both leaves and inner nodes of the different trees, leading to a good load fairness. This system therefore aims at heavy traffic data streams,

[1] Latency stretch is defined as the ratio between the latency accumulated over multiple hops from source to destination and the direct latency between source and destination.

such as media content. The trees are built and maintained as separate Scribe trees. To deal with nodes whose out degree is exceeded, SplitStream adds a child rejection mechanism. For children that find no new parent, SplitStream maintains a spare capacity group of nodes that can handle additional children.

A similar approach is taken by Bullet [94]. It also splits the data into disjoint encoded blocks, which are distributed via a mesh over an overlay tree. The overlay tree is constructed and maintained by any of the existing algorithms. The delivery through the mesh is probabilistic, but the block encoding scheme (e.g., using erasure codes) allows reconstructing the data even if a few blocks are missing. Disjoint content is located using the RanSub algorithm [93]. Like SplitStream, Bullet aims for high bandwidth traffic, and delivery latency is not a primary concern.

Chainsaw [128] builds on an unstructured topology and completely replaces trees with a mesh. Based on a random mesh topology, peers get notified upon arrival of new sequence-numbered packets and request ranges of sequence numbers. Unlike the previous systems, Chainsaw thus uses a pull-based approach, which is similar to the BitTorrent distribution strategy. Obviously, this concept is rather suitable for bulk data streaming than for low-latency control information dissemination.

Most of the presented approaches use node degree or bandwidth as the primary factor determining the dissemination structure. Many optimize for high throughput and assume a single source. Latency is also considered, but the average number of hops is usually too high for sub-second end-to-end latencies. They further provide no prioritization, but rather rely on a simple multicast group interface. Further, an adaptation to application needs is barely considered. A comparison of the discussed approaches based on their properties is provided in Table 3.1.

3.3 Delay Optimization in Multicast

Although multicast and in particular application layer multicast has been a research topic for more than two decades, only recently there have been new approaches dedicated to the analysis and optimization of the delay in message dissemination. An important novelty is the explicit consideration and modeling of queuing delay on each node, caused by bandwidth limitation. When a node that forwards a message to several neighbors “simultaneously”, the last of the messages may have to wait for a significant amount of time to be actually transmitted, depending on fan-out, message size, and bandwidth availability. To exemplify, a 1 kB packet takes about 12 ms to be transmitted on a 1 Mbps line. With a fan-out of 10, the last packet has a delay of 100 ms before even being put on the line. The fan-out but also the message transmission order therefore have a significant impact on the individual message delays, which has been largely ignored in the previously described works.

Mokhtarian and Jacobsen [116, 117] propose algorithms that optimize the delay of multicast trees. In their model, they include the time messages are delayed on the nodes before getting sent out. The authors consider the two problems of minimizing average and maximum latency. They show that the problems are NP-hard and propose heuristic approaches to solve them efficiently. Their first paper [116] only covers constructing whole trees, incremental updates are added in

Approach	Single- vs. multi-source	Tree construction	Considered node/ network features	Delivery	Adaptation (environment)	Adaptation (application)	Failure fall-back
Overcast [83]	single	receiver	node bandwidth	deterministic	receiver-based relocation	no	receiver-based relocation
BTP [77]	multi	distributed	node degree, node-to-node latency, tree cost	deterministic	sibling switching	possibly	re-connect to root
ALMI [130]	multi	central (session controller)	node-to-node latency	deterministic	central recomputation	no	central recomputation, member- or controller-initiated
NICE [15]	single	distributed, cluster-hierarchy	node-to-node latency, node degree	deterministic	cluster-based refinement	no	cluster-based
OMNI [16]	single	distributed (among MSNs)	latency, node degree	deterministic	distributed (among MSNs)	no	none / move one child up the tree
Scribe [33]	multi (single rendezvous)	DHT-based (Pastry)	Pastry locality (latency)	deterministic	distributed, Pastry locality (latency)	no	DHT-based (Pastry)
SplitStream [32]	single	multiple Scribe trees	Pastry locality (latency), node degree	deterministic	not beyond Pastry	no	Scribe, coding fault tolerance
Bullet [94]	single	mesh on tree	number of senders/receivers	probabilistic	distributed periodic re-evaluasion	no	no specific, coding fault tolerance
Chainsaw [128]	single	random mesh	bandwidth	deterministic	no	no	multi-source pull
Narada [41]	single	distance vector spanning tree on mesh	bandwidth, latency	deterministic	continuous mesh optimization	variation of latency-bandw. tradeoff	neighbor failure detection, partition detection & repair
Mokhtarian 2013/14 [116, 117]	single	source	node bandwidth, inter-node delay	deterministic	no	min-avg-lat vs. min-max-lat	source-based
Li 2013 [108]	multi	central with distrib. heuristic update	node bandw. (packets), inter-node delay	deterministic	distributed refinement	no	possibly distributed
Streamline [113]	single	source	node bandwidth	deterministic	tree recalculation on changes	no	new parent selection with global knowledge

Table 3.1.: Comparison of a selection of application layer multicast algorithms

their follow-up version [117]. The trees are calculated at the source nodes and encoded in the data packets, so that intermediate nodes do not need to keep state. Additionally, this gives the source node the best possible flexibility in selecting the tree structure. While in certain cases the path information can be encoded economically using bloom filters, other cases require an overhead of up to $O(n)$ with n being the number of nodes in the tree, which is significant. For small payloads, this option seems infeasible. Furthermore, the information each node needs to exchange with others to run the optimization algorithm is $O(DLN)$, where N is the total number of nodes, L is the pairwise shortest path hop count, and D is the node degree. Especially the linear dependency on the network size N is critical for large networks. For small receiver sets compared to the network size, there might be an option to prune the information without overly restricting the optimization process. Finally, all intermediate node or link failures must be quickly reported to the source node because only the source node is able to rebuild the tree. Other intermediate nodes that are first notified about the failure are therefore unable to react independently.

A very similar investigation was performed by Li et al. [108]. They also argue that the delay incurred on the nodes due to queuing is a significant source of latency and must be included in the optimization model. The authors distinguish between transmission and waiting time and model waiting time using queuing theory. They define two problems to be solved incrementally. The first is to build a multicast tree for each peer so that all receivers can be reached and total (i.e., average) delay is minimized. The second is to find multicast trees that reach all receivers and satisfy node bandwidth constraints. NP-completeness is proven for both problems. The main algorithm is designed to run on a central server, which needs to know all nodes' capacities and latencies and which disseminates the results to the nodes. The authors do not explain what information each node needs to perform the forwarding and they do not analyze the necessary communication overhead. To add or remove nodes from an existing multi-tree, the authors propose distributed operations that can be initiated by all nodes. The same holds for a refinement operation that is supposed to react on environment changes. It remains, however, unclear what exact information each single node needs to perform these operations and how the information is exchanged. Furthermore, distributed operations and updates are barely evaluated. In a follow-up paper [109], Li et al. present a genetic algorithm for the central multi-tree computation.

3.4 Publish/Subscribe

A main motivation for the publish/subscribe paradigm is the decoupling of sender and receiver, as pointed out by Eugster et al. [54]. In its core concept, participants that are interested in a particular kind of information can *subscribe* to receive corresponding *event notifications*. The senders (*publishers*) do not need to know their receivers; this is transparently handled by the publish/subscribe system, often implemented as a publish/subscribe *middleware*. The middleware might also be able to store events for subscribers that are offline at the time a message is published. This concept decouples senders (publishers) and receivers (subscribers) in space and time. Furthermore, publishers can pass events asynchronously to the middleware and therefore do not need to handle or care about the delivery process (synchronization decoupling [54]).

Publish/subscribe systems are commonly distinguished by the way they express *interest* in subscriptions. The most basic option is *channel-based* publish/subscribe, providing fixed channels in which events are placed and which can be subscribed. Conceptually, this is similar to group-based multicast. A slight extension of channel-based publish/subscribe is *topic-based* publish/subscribe, which allows additional filters on message header fields. *Subject-based* publish/subscribe provides a hierarchically structured name space and allows subscriptions on different levels on the hierarchy. For example, the hierarchy "news/sports" can be subscribed as "news/sports", or less selectively as "news", which could also include events on "news/business". The most expressive scheme is *content-based* publish/subscribe. It allows subscriptions with a logic term of predicates based on event properties, such as "type = 'sensor' AND temperature > 30". Finally, special purpose publish/subscribe types provide subscription criteria tailored to certain kinds of applications. A prominent example is the class of spatial publish/subscribe systems, which is implemented by many of the virtual reality overlays presented in Section 3.6.

The degree of expressiveness has an impact on the performance and/or the cost of the system. While channel associations can be matched with little effort, especially if the number of channels is limited, content-based publish/subscribe requires matching each event against each subscription individually in the extreme case. Specialized types (e.g, spatial publish/subscribe) have the potential to be more efficient because they reduce the subscription space while at the same time fitting the application needs, but on the other hand are limited to a smaller number of applications.

Traditional distributed publish/subscribe systems use broker-based architectures. All participants connect as clients to one of possibly multiple brokers, which distributes the client's publications and notifies the client based on its subscriptions. If the system consists of more than one broker, the brokers build a network, forming an overlay topology.

Essentially, a publish/subscribe system fulfills two main purposes: *filtering* and *multiplication* of events. Event filtering makes sure that clients only receive the events that match their subscriptions and therefore minimizes network traffic and client load. Event multiplication is necessary because potentially multiple clients have to be notified of a single publication. In broker-based systems, these two tasks are handled by the brokers, which are running on reliable and well-connected servers.

There are several commercial implementations of message oriented middleware supporting the publish/subscribe paradigm. Some of the most prominent examples are *TIBCO Rendezvous*[2] and *Apache ActiveMQ*[3]. With *Java Message Service* (JMS) [76] and *Advanced Message Queuing Protocol* (AMQP) [161], a few standard protocols for message oriented middleware have emerged, which allow inter-operation across products. These systems aim at performance and scalability, but assume stable and managed server environments.

Some of the popular research prototypes are *Hermes* [132], *Padres* [60], *Rebeca* [120, 6], and *SIENA* [31]. They commonly employ a distributed broker architecture, building spanning trees for message distribution on arbitrary broker topologies. An important question addressed by these systems is how filters can be efficiently moved as close to the source as possible in order to filter

[2] TIBCO Rendezvous. http://www.tibco.com/products/soa/messaging/rendezvous/default.jsp
[3] Apache ActiveMQ. http://activemq.apache.org/

out messages as early as possible, saving communication overhead. *Filter merging* minimizes the number of filters at the brokers to reduce computation effort.

All conventional broker-based solutions require dedicated infrastructure, which has to be managed and maintained, inducing non-negligible costs to run the system. Additionally, requirements for low latencies demand optimized broker and subscriber placements [36]. This increases deployment costs further.

Peer-to-Peer Publish/Subscribe

Peer-to-peer-based approaches do not require dedicated brokers and instead use the participants (peers) to perform event filtering and multiplication. Hence, they waive the distinction between clients and brokers. Being peer-to-peer systems, they are based on the assumption that any participant can leave the system or fail at any time. They use mechanisms for redundancy and decentralized organization to achieve an increased resilience against failures.

There are numerous approaches to build publish/subscribe functionality on top of peer-to-peer overlay networks. An early peer-to-peer system for application-layer multicast (cf. Section 3.2) is Scribe [33] by Castro et al., based on which a simple channel-based publish/subscribe can be implemented. Scribe is built on top of Pastry [139], a generic peer-to-peer routing overlay, which provides a certain degree of resilience against failures. With Meghdoot [72], Gupta et al. present a content-based publish/subscribe approach based on the distributed hash table (DHT) CAN [135]. Also based on a DHT, in this case Chord [148], is the system Mercury [22] by Bharambe et al. Meghdoot projects its attribute space onto CAN's n-dimensional Cartesian space. Mercury uses one Chord ring for each attribute. Like Scribe, both inherit their reliability properties from the underlying DHT. Terpstra et al. [154] propose a content-based publish/subscribe solution on top of Chord as well, by combining a broker network with the DHT. For this, the broker network is extended to handle more general topologies than just trees, while maintaining its expressiveness. The DHT routing guarantees an $O(n)$ routing depth.

DHTs achieve high scalability by requiring only $O(\log n)$ (Chord, Pastry) or $O(dn^{1/d})$ (CAN) overlay hops for routing, where n is the network size and d is CAN's number of dimensions. On the other hand, when latencies matter, even $O(\log n)$ overlay hops can be far too much. Consider a Chord network with $n = 1.000$ nodes. On average, a Chord message takes $\frac{1}{2}\log_2 n \approx 5$ hops. With typical Internet latencies, a message transmission can thus easily take more than one second. DHT nodes build the topology only based on their node IDs, which does not allow clustering participants with similar interests (i.e., subscriptions) for an accelerated event delivery.

This problem is partially tackled by the two content-based publish/subscribe approaches Sub-2-Sub [163] and Mirinae [38]. In Mirinae, the peers arrange in a virtual hypercube, in which published events are routed. The position of each node in the cube is determined by its subscription predicates. Sub-2-Sub uses an unstructured network in which peers continuously exchange subscription information and form clusters based on this information. This reduces the involvement of peers in the propagation of events they are not interested in and therefore also reduces total transmission overhead. The number of hops for message dissemination, however, remains significant. In the Sub-2-Sub experiment with 10.000 nodes, many events take more than

ten hops to be disseminated to all subscribers. At the end of the day, this is the cost to be paid for a generic content-based publish/subscribe.

Another approach to generic publish/subscribe on peer-to-peer overlays is to use rendezvous systems [105] like BubbleStorm [153]. Rendezvous systems allow for an efficient and scalable search based on arbitrary attributes and in an analogous manner publish/subscribe functionality based on arbitrary matching criteria. A lookup or subscription matching, however, still requires multiple hops to reach $O(\sqrt{n})$ nodes in the system. While this works well enough for a timely neighbor discovery [112], it would induce too much overhead and dissemination latency, if used for the propagation of all regular update messages.

Subscription Dynamics

For applications with dynamic interests, such as virtual worlds with moving avatars, an additional challenge for publish/subscribe is the regular update of subscriptions. Performing full re-subscriptions can be very costly as filters may have to be re-optimized each time. This issue is addressed by *context-aware* and *parametric* publish/subscribe. Context-aware publish/subscribe, as described by Cugola et al. [46], explicitly takes into account the "situation in which the information to be communicated is produced or consumed." [46] An example for this situation (i.e., context) is the positioning of publishers and subscribers. The authors argue that keeping such information in a separate context is preferable to encoding it into the publications and subscriptions for different reasons. Beyond the filter update efficiency indicated above and a better separation of concerns, the main argument is the matching inversion that is necessary to model cases where events should have a limited range: In such cases, the position of the subscriber, i.e., data, needs to be encoded in the filter condition. This conflicts with the original concept that subscriptions do not hold data, but only constraints on the data.

Parametric publish/subscribe [84] is a further generalized variant that introduces parameters in subscriptions that can be updated separately, avoiding frequent re-subscriptions. Additionally, using approximation techniques, the system can mitigate parameter update rates that are too high to be handled directly. As above, examples for such parameters are dynamically changing coordinates, but also target prices in trading systems.

Especially the general parametric context-based publish/subscribe is a powerful and highly expressive model for dynamic subscriptions. Being a general-purpose solution, however, parametric publish/subscribe still requires the parameters to be updated, and thus communicated, directly. In cases where the main or the only subscription criterion depends on parameters, the optimization potential shrinks. This is where application-aware solutions, with constrained functionality but with more knowledge about data semantics, can bring further improvements.

3.5 Adaptive Overlay Topologies

There are a few approaches to build adaptive overlays with different dissemination strategies. They explicitly focus on changing conditions, but apart from that follow the concepts of application layer multicast.

With *Hyphen* [3], Allani et al. present a middleware protocol for a generic construction of overlay trees. Targeting ALM, content distribution, and media streaming, it is designed for single-tree, multi-tree, and mesh topologies. The topologies are constructed using gossip messages in the peers' local neighbor sets, which are selected based on the Streamline [113] concept. Topologies are constructed by forwarding control messages from the source and by selecting the best paths according to a generic path quality function. The path quality can be composed from the three components bandwidth, latency, and reliability. Peers are limited to connections to neighbors from their local neighbor set. Therefore, the algorithm cannot leverage low-latency and high-bandwidth links between peers, unless they are represented in their neighbor sets.

Streamline [113], proposed by Malekpour et al., decomposes the data dissemination problem into two parts, which are implemented as two layers. The first is the construction and maintenance of a mesh overlay. Using a gossip-based protocol, starting from their local view, nodes select peering nodes (neighbors) based on a selection function that keeps the node degree between a certain minimum and maximum. The second layer builds an optimized diffusion tree for data propagation. Tree construction is based on a global network view including each nodes' peer set and bandwidth that each node gathers using a gossiping mechanism. The reliability of updates is estimated using a distortion factor that is increased in each gossiping step. The multicast source node uses the global network view to build an optimized propagation tree. Each newly calculated tree is embedded in full in a multicast message to be disseminated to all relevant nodes. To react promptly to node departures and failures, a node in the tree detecting the departure of its parent searches for an alternative parent based on its global network view and the tree information it has received.

Although the gossiping reduces the necessary overhead compared to a full broadcast of updates by trading off accuracy against overhead, maintenance of global state severely limits the scalability of this approach. Furthermore, piggy-backing the full propagation tree on each change causes a high overhead in dynamic networks.

3.6 Interest Management and Application-tailored Multicast

For the purpose of enabling peer-to-peer multiplayer online games (MMOGs), a significant amount of research has been conducted to find bespoke solutions. In many cases, the aspect of event dissemination is tightly coupled with *interest management*. Interest management determines which participant requires which part of the game world information. This function is necessary whenever the total number of participating players or world size do not allow sending updates for the full state to all players, which is the case for MMOGs. Usually, the basic concept of interest management is the *area of interest*, which is bounded by the player's vision range. Assuming an omnidirectional vision range, the area of interest is often treated as a circle around the player in the virtual (2D) game world.

The event dissemination within each player's vision range can be translated into multicast group memberships or receiver sets. By combining the two aspects, efficiency gains can be obtained, e.g., by sharing neighborhood information. Furthermore, the event dissemination can be fit to special needs, such as different priority levels. On the downside, however, a tight coupling reduces

flexibility because it becomes harder to exchange single components, e.g., only the dissemination algorithm.

In the remainder of this section, we look at such approaches. Section 4.1 provides further discussion on the composition of interest management and event dissemination.

DHT-based Online Gaming

Soon after the emergence of distributed hash tables (DHTs), which pushed the peer-to-peer paradigm forward in the early 2000s, steps were taken to use DHTs for enabling MMOGs on a peer-to-peer basis.

One of the first is SimMud [91] by Knutsson et al. The game world is split into regions whose size is based on the players' vision range. Interest management is performed by distributing information only within a region. Thus, each region forms an interest group. Player's positions are multicast in fixed intervals to all other players within the region. SimMud also supports mutable objects. Each object is assigned to a coordinator, a node that is responsible for resolving conflicts when the object is manipulated. Updates to objects are multicast to all players in the containing region. The system builds on top of the Pastry DHT [139]. Each region has an ID assigned and is mapped onto the DHT. The node responsible for a region's ID is responsible for that region. To achieve a better load balancing when certain regions are under heavy load, the authors propose to create a different region ID for each object type. Fault tolerance is achieved by replicating the game state objects over several nodes so that one of the secondary copies can take over when the primary copy holder fails.

As discussed for DHT-based publish/subscribe systems (Section 3.4), DHTs are not well suited for use cases where a low update dissemination latency is of importance. DHT-based games are therefore limited to slow-paced games where latencies in the order of one second are acceptable. Further, fixed-size regions like those used in SimMud have the downside that there remains the trade-off between small regions that have to be switched frequently and large regions that become overloaded in crowded places. Dynamically adjusting region sizes can be complex and expensive, depending on the used DHT. These problems are tackled by the dynamic overlays that we discuss in the following. They build the topology based on the virtual world coordinates, so that close players have direct overlay links. This reduces routing overhead and message dissemination delays.

pSense

pSense [143] is an unstructured and dynamic peer-to-peer network specialized for MMOGs. Based on locality within the game world, it provides interest management and efficient multicast in a highly dynamic game world. Unlike many other peer-to-peer gaming approaches, pSense does not split the world into static regions. Instead, every node (i.e., player) keeps track of its neighbors in the game world proximity and only exchanges information with those. Therefore, each node is immediately informed about activities nearby and almost completely unaware of everything that happens outside its vision range.

The main task of the pSense topology is to keep the network connected, as there is no underlaying system like a DHT or super peers with global knowledge to do this. Each node maintains two lists, a *near node list* and a *sensor node list*. The near node list contains all nodes in the circular

vision range. pSense assumes that vision ranges are symmetric, i.e., that all neighbors in a given node's near node list have the node in their list. The sensor node list contains nodes that are just outside the vision range. The area around the vision range is partitioned into eight sections for the different directions (Figure 3.1a). In each of those sections, the node closest to the vision range (but still outside) is selected as sensor node. The sensor node list is maintained by periodically sending sensor node request messages to all nodes in the sensor node list, returning new sensor node suggestions. When joining the network, the new node connects to some arbitrary node that is already in the network. That node provides information about nodes closer to the joining node via a sensor node request. The new node then repeatedly sends sensor node requests, and its position updates are forwarded by the sensor nodes, so that the node eventually finds all vision range neighbors.

Messages (position updates) are sent directly to all nodes in near node and sensor node list. If the outbound bandwidth capacity is exceeded, a random subset of messages is dropped. To make sure that all nodes in the vision range receive the message, receivers forward messages to all known nodes which are within the vision range of the originator but not in the message's receiver list, as long as a certain hop count has not been exceeded.

pSense's interest management scheme is simple but effective and has shown to be stable in highly dynamic situations [69]. Yet, it is in its original version not prepared to be adaptive to changing loads. While player discovery works well in crowded environments, the direct transmission of events between players can exhaust the available bandwidth. We have shown [69] that pSense's forwarding scheme quickly exhausts the participants' node bandwidths. One of the reasons is that the whole receiver set has to be sent with each message. This is particularly expensive for small payloads, even with moderate receiver set sizes. Therefore, an improved and adaptive event dissemination system can be a highly beneficial complement.

nSense

nSense [137] is a proposed extension to the pSense overlay that aims at generalizing interest management to higher numbers of dimensions. The circular area of interest becomes an n-dimensional ball, whose $(n-1)$-dimensional sphere is divided into a variable number of sectors. Beyond a better selectivity for three-(and even higher-)dimensional game worlds, this can be used to provide visibility layers, wormhole portals, or the like. The event dissemination mechanism, however, remains unchanged and thus has the same shortcomings as pSense's.

VON

The core idea of VON [78] is to build a Voronoi diagram [71] from the players' positions, which are scattered on a 2D plane, the NVE/game world. Using the Voronoi diagram, each node can find all *enclosing* and *boundary neighbors*. Enclosing neighbors are nodes that have a common edge with the node in the Voronoi diagram. Boundary neighbors are those whose areas are cut by the node's area of interest border. The Voronoi diagram is not built globally, but every node builds its own for its proximity (Figure 3.1b). Position updates are sent to all recorded neighbors in the Voronoi diagram. Boundary neighbors inform the node when there is a new node overlapping the area of interest boundary. For joining the network, the new node contacts some node in the network

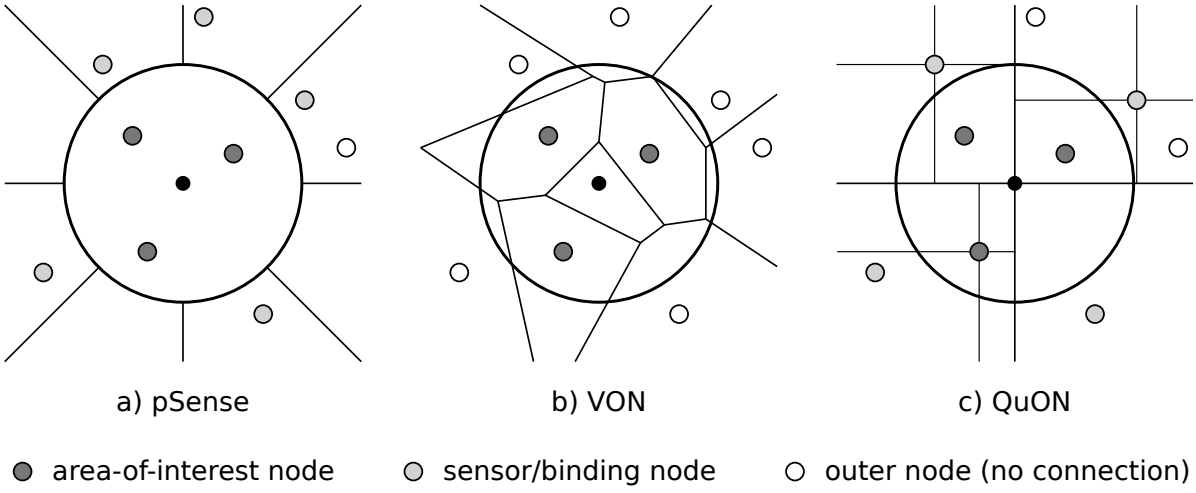


Figure 3.1.: Typical decentralized interest management topologies: VON, pSense, and QuON. Apart from the different ways of structuring neighbors, they work very similarly. All use a circular area of interest and have direct connections to all neighbors within the area of interest. Connectivity of the overlay is maintained using sensor nodes (pSense) or binding nodes (QuON). VON only uses inner-area-of-interest nodes to inform about new neighbors based on the Voronoi structure. With up to eight sensor nodes, pSense requires the most extra connections to nodes outside the area of interest. This increases traffic, but also improves robustness.

who forwards the join request to the node whose region contains the desired position of the new node. This *acceptor node* sends the list of neighbor nodes to the new node which then builds up its Voronoi diagram. Before leaving the network, a node informs all of its enclosing neighbors, which may then decide to notify neighbors about changes where necessary.

Messages are transmitted directly from the sender to all enclosing and boundary neighbors. To avoid bandwidth bottlenecks in crowded areas where nodes have many neighbors, the authors propose a dynamic area of interest adjustment. Nodes shrink their area of interest radius when a certain connection limit is exceeded and restore it when the number of connections falls below that limit. This adaptation concept on the interest management layer effectively treats outer neighbors in the vision range with lower priority.

Voronoi diagrams allow for efficient and completely decentralized overlay topology management. There is no global knowledge necessary, and nodes only communicate with neighbors in their vision range. Direct communication causes low delays and thus a good awareness of the surrounding players. However, the Voronoi diagrams need to be maintained. Especially in very crowded places this causes a high transmission and computation overhead as well as topology inconsistencies due to the frequent changes of the Voronoi diagram [69]. Direct communication may overburden the connection capacities of some nodes. A dynamic area of interest limits the required bandwidth, but this will not help for heterogeneous networks as a node cannot disconnect from nodes whose area of interest cover its position. Message forwarding approaches such as described in this work can help overcoming this problem.

QuON

QuON [13] follows an approach similar to pSense and VON, but uses point quad-trees to structure neighboring nodes in the virtual space. Each participant builds a local point quad-tree centered at its own position. Known neighbors are sorted into the four regions, each one splitting the region into four sub-regions (Figure 3.1c). Like pSense and VON, QuON maintains a set of direct neighbors that are located in the area of interest and to which messages are sent directly. To maintain connectivity, one *binding neighbor* outside of the area of interest is selected per quadrant. Neighbor discovery works similarly to pSense. Upon movement, all neighbors notify about new neighbors that are located in the new but not in the old area of interest. Additionally, the list of binding neighbors and their positions is sent to all binding neighbors so that they can suggest better binding neighbors. As a backup mechanism against inconsistencies due to delayed messages and failures, QuON increases the effective area of interest by a certain factor and therefore stays connected to a larger number of direct neighbors.

Like pSense and VON, QuON lays the focus on an efficient distributed interest management. The dissemination of messages within the area of interest using direct communication, however, is as simple as pSense's and VON's with the discussed drawbacks.

Donnybrook

Donnybrook [21] by Bharambe et al. uses the idea that in multiplayer games, a coherent view on the virtual world is not necessarily needed. Instead of sending high precision updates to all players in vision range, Donnybrook's interest management identifies the five most relevant neighbors for that player, the player's *interest set*. This approach is based on the insight that humans can only concentrate on a constant number of objects at once—studies found numbers between 4 and 7 [45, 138]. The importance of a particular neighboring player is based on three factors: proximity, aim, and interaction recency. Interest set neighbors are subscribed for high frequency updates, while the remaining neighbors only send *guidance* information once a second. This guidance is used by artificial intelligence players (bots), the so-called *doppelgängers*, that run on each player's machine to approximate the neighbors' actions. The purpose of the doppelgängers is therefore to make the neighbors that are not focused by the player look plausible, even if they do no behave identical to their real counterpart.

Donnybrook has two different message dissemination schemes for update messages and for guidance messages. Interest sets limit the traffic caused by incoming updates, however, there may be nodes that are of interest for many others but whose outgoing bandwidth does not suffice for satisfying subscribers. To mitigate this problem, nodes with spare capacity can help disseminating these updates. If a node's available bandwidth is greater than a certain reserved amount, it joins a *forwarding pool* serving those with scarce bandwidth. Nodes whose outgoing update messages exceed their available bandwidth build a probabilistic forwarding tree by choosing a random subset from the forwarding pool, whose total capacity suffices the requirement. This is done separately in each frame where necessary. As the selection from the forwarding pool is made randomly, the nodes in the pool advertise only half of their capacity to be able to cope with being selected twice in one frame.

Guidance messages of nodes with a low outgoing bandwidth are also forwarded by a set of forwarder nodes. The assignment of these forwarder nodes is done at the beginning of the game. Guidance forwarders are nodes that are not in the update forwarding pool and have spare inbound capacity. Depending on that spare capacity, a node can forward guidance messages for a certain amount of other nodes. The guidance messages are always sent to all other players. Nodes forwarding guidance information for multiple nodes may reduce traffic by aggregating guidance messages. Failed forwarder nodes are detected via timeouts. Replacement forwarders are discovered using a bit in each guidance packet, indicating that the sending node still has unused forwarding capacity.

The principle of Donnybrook's interest sets includes game semantics to estimate the importance of certain objects to a particular player. This is a very powerful feature effectively reducing network traffic without compromising the user's view of the game. On the other hand, this feature requires specific adjustments for each game, maybe even for each gameplay mode. Replacing less important players with doppelgängers is an advancement of the dead reckoning [129] technique. Its strength, of course, depends on the predictability of player actions. Reducing the high fidelity interest set to a fixed size of five is only feasible assuming that the predictions on the importance of the individual players is very reliable and that five is sufficient even for experienced players. There is evidence that playing video games can increase the capacity of tracking multiple objects [158]. Further, Donnybrook's messaging scheme, which disseminates guidance info to all players, is not particularly scalable. The same applies for the static guidance forwarding and per-message probabilistic forwarding. We believe that Donnybrook could also benefit from using InterestCast as the message dissemination layer.

4 Approach and Design Decisions

In this chapter, we highlight the major design decisions for the development of InterestCast. By discussing possible options and their consequences, we argue why we choose certain options and thereby lay the foundations for the design of InterestCast. We follow a top-down approach, starting with the fundamental architectural issues and move towards the more technical and implementation details.

We begin with a decomposition of functionality into the two components *interest management* and *event dissemination* and discuss their interfaces. We afterwards dive into the topic of adaptability, first presenting universal concepts with a focus on utility functions and subsequently narrowing down to the application for our use case of event dissemination. In the third part, we take a closer look at topology construction and event routing schemes. After that, we derive an optimization strategy based on the discussed adaptability concepts. Finally, we identify an optional extension for using dedicated helper nodes supporting the operation.

4.1 Decomposition and Interfacing

The first aspects to discuss here are the decomposition of the application level protocol stack and the interfacing between the components and the application. The goals are to maximize *flexibility* with respect to application demands, *reusability* of protocol components, and runtime *efficiency*, which includes minimization of protocol overhead and provision of adaptation capabilities.

As introduced in Section 3.1, the publish/subscribe paradigm provides a good abstraction for a decoupled event dissemination to the application. In particular, many of the applications considered in this work require a spatial publish/subscribe model, e.g., with respect to the virtual or physical world. Although conceptually similar, the applications have differences in their models of representation and therefore need slightly different publish/subscribe filter criteria to fit their native representation. One example for such case is the number of dimensions of the virtual world. Overlays for virtual environments usually assume a 2D world and therefore only support 2D coordinates, because most games have a 2D or quasi-2D world. There are, however, good reasons to allow more than two dimensions, e.g., for full 3D or even 4D space simulations[1], fast portal effects, or game unit type layers [136, 137]. Another example is the asymmetry between longitude/latitude and height in air traffic. Height is measured in much smaller units because of the thinness of the atmosphere, and the space is vertically separated in a different way than horizontally [70].

A generic content-based publish/subscribe solution can in principle cover all those cases. Such approach, however, would at least harm efficiency. First, the subscriptions might not be defined as precisely as needed by the application. For instance, circular range-based subscriptions might

[1] Examples of 4D games are Adanaxis [122], a space shooter in 4D euclidean space, and Miegakure [152], a puzzle game in a 4D block world.

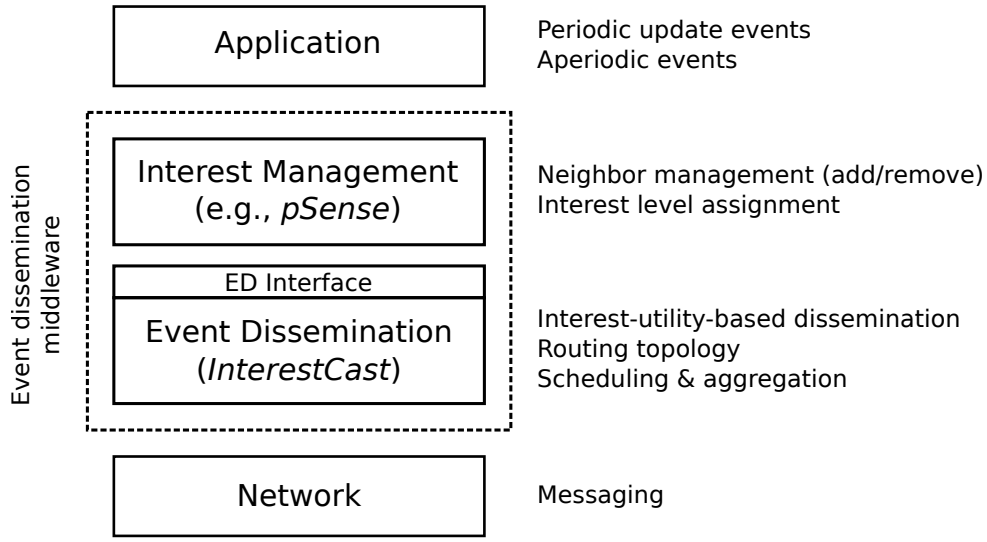


Figure 4.1.: Layering of the basic components in the interaction with InterestCast. Together with an interest management implementation such as pSense, InterestCast provides the event dissemination middleware.

have to be extended to a rectangle to be specified in a way compatible with attributes used in content-based publish/subscribe. Such reduced precision increases the amount of false positive matches and therefore introduces overhead through unnecessarily delivered events. Second, filtering and delivery can be more effectively optimized the more domain knowledge is available in the middleware, e.g., by prioritizing events using implicit application knowledge or making domain-specific assumptions about the need for certain events. This is particularly the case for distributed peer-to-peer solutions.

We therefore follow a more specialized approach. As a common abstraction, we use *interest*. Entities, such as users and possibly other application objects, have interest in each other. From the generalization of peer-to-peer gaming overlays [56], we adopt the aspects of interest management and event dissemination. Unlike in most existing gaming overlays, however, we explicitly separate the two concerns to provide both flexibility and reusability.

4.1.1 Decomposing Interest Management and Event Dissemination

For a decomposition, we need a clear separation of concerns as well as an interface between the two components. The first aspect is subject of this section, the second is approached in the next section.

Figure 4.1 illustrates the basic composition and interaction of interest management and event dissemination, which together provide the application-specific event dissemination middleware. The application-specific parts are contained in the interest management part, whereas the event dissemination part, which is provided by InterestCast, is reusable for a wider range of applications.

Interest Management

Interest management determines which entities are of interest for a given entity. An entity interested in another entity has to receive state updates of that other entity. State can include position, battery life, current task, etc., depending on the application. There might be multiple classes of state with different interest assignments. In our consideration, without loss of generality, we assume a single state type. Multiple classes can be implemented, e.g., by using multiple interest management instances in parallel. Furthermore, in the following, we assume a single entity (e.g., user) on each node. Multiple entities per node can be achieved by introducing virtual nodes, one for each entity.[2]

Interest can be binary, meaning that an entity is either of interest or is not. This corresponds to a regular subscription. In such case, all interested entities are handled the same way. Interest, however, can also have gradations. The *interest level* quantifies the degree of interest and therefore the importance or urgency of state updates. This is particularly useful where some nodes might not have enough capacity to serve all interested nodes with the highest fidelity. Interest levels allow prioritizing update delivery. Donnybrook [21] (cf. Section 3.6), for example, distinguishes two classes of update fidelity, corresponding to three interest levels (none, low, high). Instead of providing two separate classes, the factors that Donnybrook uses for classification (distance, direction, interaction) can likewise be used to determine a *continuous* interest value. We therefore model interest as a continuous value to cover all of the described options.

In addition to interest level assignment, interest management is also responsible for managing the set of neighbors that are of interest or interested, respectively. This particularly includes adding new neighbors of interest. The need for this arises from the fact that an interest value can only be assigned to existing neighbors. In the simplest case, all nodes in the system are added as neighbors, with only a subset having an interest value of greater than zero assigned. Maintaining connections to all other participants, however, imposes a serious limitation to the scalability of the system. It is, therefore, the task of the interest management to preselect the set of interested participants so that each node has to maintain connections to only a subset of all nodes. This selection can possibly be done by a central instance that is informed about all participants' positions. Alternatively, a more decentralized approach is to agree on fixed regions based on which participants can form groups. This is a common technique also for server selection in large game worlds, such as Second Life [111]. More fine-grained distributed interest management specifically for virtual environments in an euclidean 2D space is provided by the overlay algorithms VON [78] and pSense [143] (cf. Section 3.6).

The interest management scheme in many cases depends on the application or application type. This manifests itself in the multitude of overlays that are designed specifically for games and virtual environments. Generic solutions, e.g., using general-purpose attributes and matching criteria as seen in content-based publish/subscribe solutions, can fit the needs of such applications as well. More specific solutions, however, can improve performance and/or efficiency. A good example is the euclidean space, which can be modeled with coordinates being represented as generic at-

[2] Using virtual nodes can lead to unnecessarily receiving redundant messages for multiple entities. Using an appropriate deduplication mechanism as we will introduce for InterestCast, however, duplicates can be effectively avoided.

tributes. A selection for a rectangular (more generally: orthotopic) subspace can be formulated as a simple n-dimensional range query (e.g., $x \geq x_A \wedge x \leq x_B \wedge y \geq y_A \wedge y \leq y_B$). When circular shapes are to be selected for a direction-independent constant vision range, more complex selection operations are required (e.g., $x^2 + y^2 \leq r^2$). Such queries are often not supported or hardly efficient to implement generically in a distributed fashion. An option is to use bounding rectangles (othotopes), but this implies false positive results, in turn reducing efficiency. The higher efficiency can therefore be reached with interest management solutions designed for a certain domain.

To sum up, the interest management part is responsible for discovering neighbors of interest and assigning interest values to the neighbors. The rules based on which this is done are mostly application-specific and can therefore vary.

Event Dissemination

The purpose of the event dissemination part is to disseminate the events generated by the applications to the interested neighbors according to the assigned interest levels. Unlike interest management, the event dissemination is mostly application agnostic.

Since interest management primarily takes application needs into account, the event dissemination has to serve as a moderator between the application needs and the network properties. The latter include first and foremost the available bandwidths as well as the network latencies between the nodes. Hence, the event dissemination should optimize to fit the application needs to the underlying network reality as well as possible. This includes the selection of the network paths on which event messages are transmitted, their aggregation, and scheduling, among others. By abstracting the application needs, the event dissemination can perform this self-optimization without built-in application domain knowledge.

Many existing works have a very primitive event dissemination mechanism strongly coupled to the interest management. Server-based approaches, including publish/subscribe broker networks, assume that the servers have sufficient capacity to aggregate information and send it to each of its clients individually. In most peer-to-peer-based approaches (cf. Section 3.6), peers send all their updates to their neighbors directly. Capacity shortages are mitigated with simple forwarding techniques, e.g., probabilistic forwarding (pSense [143]) or global forwarder pools (Donnybrook [21]). For this reason, InterestCast focuses on the event dissemination part.

4.1.2 Interfacing

The interface between interest management and event dissemination should on the one hand be specific enough to incorporate the benefits that the integrated solutions (pSense, Donnybrook, etc.) provide, and on the other hand should be generic enough to be application agnostic. Application agnostic means that it is not dependent on the representation of the virtual space and the topology or neighbor selection strategy, respectively.

The main abstraction that InterestCast introduces for this purpose is the concept of *interest levels*. Interest levels are assigned by participants to other participants and express their interests in each other. Interest levels are normalized to the interval $[0, 1]$. A value of 0 indicates no interest, i.e., no information has to be delivered from the respective neighbor at all, and 1 indicates maximum

interest. A threshold greater than zero can be defined for the minimum interest that is necessary for delivering information from a given neighbor. With the participants as nodes and interest levels that participants have assigned to each other as directed edges, it is possible to derive the *interest graph*. In this model, interest is only defined among participants. Applications requiring interest-based event dissemination from objects that are not immediately assigned to a participant, can implement those as virtual participants in the network [115].

The assignment of interest levels can be done by the interested node, by the node of interest, or both. When the interest level is set by the interested node, we call it a *subscription*, since the receiver-based determination of interest corresponds to the publish/subscribe model. The receiver often has the best knowledge about its interest at a given time. Hence, this is the most precise interest determination. On the downside, an explicit subscription requires dedicated communication. This is particularly costly when subscriptions, i.e., interest assignments, are updated frequently. Given that using application-specific knowledge, the node of interest (i.e., the sender node) can estimate the interest level, it is possible to react more quickly on interest changes. Additionally, this reduces the necessary amount of subscription messages. To achieve this, the application can provide *interest hints* with interest estimates for neighboring nodes.

Whenever a node has an update on its state to be delivered to interested neighbors, it invokes the *disseminate* function. This function corresponds to the *publish* function in publish/subscribe. Dissemination of the update event is then performed according to the current interest- and utility-based routing configuration. For application state that is updated continuously, such as positions of moving objects, the update dissemination frequency has to be limited to some maximum value. This can be done by the application. The application can also couple the frequency to a dead reckoning scheme [129], with predicts future positions to show a smoother movement. Alternatively, InterestCast provides a pull-based dissemination that automatically retrieves the current state from the application. This option allows for potential further optimization, such as a fine-grained adaptation of update rates to the current network availabilities or piggy-backing updates for a better utilization when there is a message to be transmitted anyways. Pull-based dissemination, however, cannot take into account application-specific information that is used for dead reckoning. In addition to the dissemination function, messages can be sent to single nodes individually. This point-to-point communication can be useful when information is only relevant for a single neighbor. Instead of being sent directly to the destination, however, point-to-point messages also use the current routing configuration. This way they are piggy-backed with other data where possible.

Hence, the interface between interest management and event dissemination ('ED Interface' in Figure 4.1) has to provide for the following main functions:

Neighbor introduction The event dissemination component is notified of new neighbors of interest by the interest management component.

Interest assignment For each neighbor, the interest management component assigns and updates interest values. This can be done in both directions: by the interested node (subscriber) and by the node of interest (application-specific interest estimation). The removal of obsolete neighbors is handled by the event dissemination component autonomously based on the interest assignments.

Event dissemination Periodic update events are pushed by the application or interest management component or regularly pulled by the event dissemination component. Aperiodic events must be pushed. Further, the same event dissemination mechanism can be used to send events to neighbors individually. The interest management component can use the event dissemination for its own purposes. It is responsible for dispatching incoming events either for its own use or deliver them to the application.

The implementations of the interest management and event dissemination components together provide the event dissemination middleware.

4.2 Adaptability

A major contribution of this work is the adaptivity of the event dissemination middleware. Adaptivity relates to two aspects. First, there is adaptivity with respect to changes in the environment. In a networked application, the most important environmental factors include node capacities (throughput, processing power, memory) and properties of their inter-connectivity, such as network bandwidths and latencies. These factors effectively determine the cost of network operations, such as transmitting a network packet. Second, there is adaptivity with respect to the application. This includes both static properties and requirements, which only change when the application is replaced by another, but also workloads and requirements that change dynamically during runtime. Considering a multiuser virtual environment, such as an online game, the workload changes with the number of participants or the degree of interactivity. Requirements can change depending on the users' activities. For instance, the direct interaction between users involves tighter latency constraints than users just moving by each other. Figure 4.2 illustrates the two aspects of environment and application influences.

4.2.1 Network Conditions

Network conditions can be obtained by the middleware via measurements and from the operating system or other information services. Information that we consider here are underlay network latencies between pairs of nodes and individual node's link capacities. Network latencies are constantly measured as round-trip times (RTT) as two nodes communicate. Without a globally synchronized time, it is impossible to measure one-way latencies, so we assume symmetric one-way latencies of $\frac{1}{2}$ RTT.

The second important parameter is node and link capacities. On the network layer, each link and each node (both end and intermediate nodes) have limited capacities with respect to maximum throughput. Therefore, each network element can be the point of congestion, and the effective throughput limits may vary greatly between each pair of nodes. In most real-world cases of nodes connected to the Internet, however, the last hop to the node, usually using DSL, cable, UMTS, or LTE technology, is the throughput bottleneck rather than the Internet backbone. With this assumption, modeling and collection of capacities is simplified significantly.

Besides latencies and bandwidth limits, which imply costs from the application point of view, one can also consider costs for the transmission of data between two nodes more explicitly. Those costs

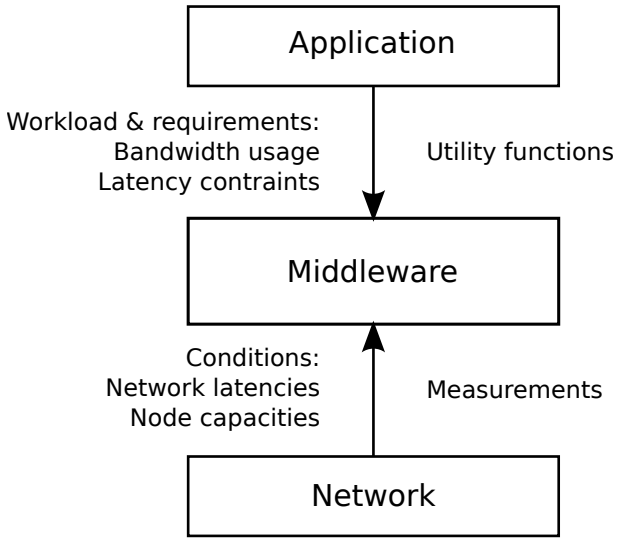


Figure 4.2.: Both the application and the network environment influence the conditions to which the system has to adapt. Environmental conditions are measured, whereas the application requirements are passed as utility functions.

could be obtained using a dedicated service. One such service is proposed by the ALTO [2] protocol, which allows querying information on network properties, such as topology and network path preferences. Availability of ALTO services, which are supposed to be provided by ISPs, however, will not be available for regular end-users and their applications in the foreseeable future.[3]

We therefore stick to the two metrics that we can measure, latency and bandwidth for the adaptation decisions. If additional information is available, however, it can be considered with an extension of our approach. More details on the measurement of network properties and conditions can be found in Section 6.4.

4.2.2 Application Requirements

While the environment (i.e., network) conditions are mostly captured through measurements, the second aspect, namely the application requirements, have to be specified more explicitly. The application workloads can be measured and therefore do not require explicit notification, unless the middleware provides means for resource reservation. Aiming for a best-effort service, we do not consider the latter option further.

For the specification of non-functional, high-level application requirements, *utility functions* have been shown to be a suitable option. In the autonomic systems community they have been used as a higher-level alternative to action- or goal-oriented self-adaptation, e.g., for data center allocation [164]. Quoting Walsh et al.: "[...] truly autonomic computing system should not require administrators to ascribe value to low-level resources. Instead, they should be able to specify utility in terms of the service level attributes that matter to them [...]" [164].

[3] Source: personal communication with Thomas Dreibholz, 2011-11-16.

Utility functions are useful wherever *trade-offs* have to be made *autonomously*. They have the advantage that requirements or desires can be expressed directly based on application-level metrics, such as response time or throughput, and conflicting needs can be purposefully traded off against each other. This unburdens the developer from explicitly specifying adaptation rules, which requires deep knowledge about the system. Utility functions do not specify hard constraints, i.e., things that must not happen such as 'no request must have a response time of more than one second'. Instead, assigning a low utility to responses taking more than a second, indicates that they have little or no value and therefore should be avoided to maximize overall utility. Hence, this representation is well suited for a best-effort service that cannot provide hard guarantees anyways.

Using utility functions for conventional Internet routing has been proposed already two decades ago by Scott Shenker [144]. This idea has been continued by Gvozdiev et al. with the system FUBAR [74], proposing flow utility based routing in software-defined networks. Both consider the two factors flow bandwidth and latency. Other approaches for using utility functions can be found in the application areas of service selection [9, 75], mobile ad-hoc networks [29, 146], and distributed systems in general [62].

4.2.3 Utility vs. Cost

Besides the utility, i.e., the usefulness of a given service to the application, it is necessary to consider *costs*. Costs in general are investments of resources. Depending on the application, use case, and particular resource, resources can be modeled as finite or infinite. Infinite resources still have a value, so that their use is to be minimized, but they do not put constraints on the possible configurations. Examples are electrical power, which may be practically unlimited, processing time, as long as there are no timing constraints, or even servers in a large data center. Finite resources could be CPU cores, memory, or network link capacities. A finite resource may be associated with no extra usage costs, meaning that using none or all that is available does not make a cost difference. In such case, the resource only implies constraints, but no costs to be minimized. As indicated, resources that are in fact limited can in certain cases be modeled as infinite to simplify the model.

There are different ways for relating the costs with utility. One option is to calculate utility and cost separately and to use the difference or quotient between utility and cost. Another option is to treat utility and costs in an integrated way, with costs being the inverse or negation of utility. This way, costs are traded off against utility in the same manner as different utilities. Further, costs can be accounted using non-linear cost functions on resource usage, like utility functions on application-level metrics. Another dimension in the modeling of utility and cost is the matter of different stakeholders, such as the service provider and the end user. In the following, we only consider utility and costs on a purely technical level, i.e., omitting monetary or user perception metrics.

The model we use in this work assumes the utility to be related to the application performance, i.e., it can be measured at the application interface. Costs, on the other hand, arise at the environment interface of the respective node, as depicted in Figure 4.3. In our particular use case, application performance is the latency of the event dissemination, i.e., the time from source to

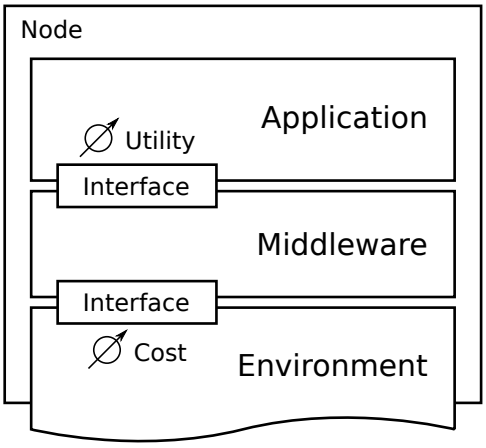


Figure 4.3.: Illustration of the generic utility/cost-based middleware adaptation concept. We model utility as the performance on the application interface and cost as the usage at the environment interface.

destination of each delivery. To trade off utility (latency) against cost (bandwidth demand), they must be made comparable. We achieve this by treating the costs as a negative utility and therefore unifying the concept of cost and utility functions.

4.2.4 Composition of the Target Function

For the optimization, it is necessary to combine the individual utilities for bandwidth demands and latencies to a total utility, or *objective function*. There are different ways to do this, most common are addition, multiplication, and minimum. The most important difference between them is in the trade-off between the individual utilities. For both the product and the minimum, single low utilities have a high impact on the total utility. Hence, there is a particular focus on fairness. Maximizing the minimum is also called *max-min fairness*. The downside of those objectives is that with a single low partial utility that cannot be increased, there is little or no potential to improve the overall utility by increasing other individual utilities. We therefore argue that they are unsuitable because the fulfillment of the partial needs cannot simply be shifted among each other, as it would be the case, e.g., in a single shared medium.

For this reason, we use an additive objective function. By itself, an additive objective does not consider fairness at all. Increasing an already high utility component is as good as increasing a low utility by the same amount. A consideration of fairness can, however, be added by using sublinear functions for the individual utilities. Let us assume a utility function $U(m)$, where m is a higher-is-better input metric, and an objective function $U_{\text{total}} = \sum_{m \in M} U(m)$. With U being sublinear, an increase of a low m adds more to the total utility than increasing a high m. Hence, there is a reward for increasing fairness. If m is a lower-is-better metric, which is the case for both bandwidth demand and latency, the relation needs to be inverted. $U(m)$ then has to decrease superlinearly with m.

4.2.5 Specifying and Updating Utility Functions

As discussed, utility functions provide a clear abstraction for the applications to specify their performance demands. The concrete utility functions are specified, and possibly updated during runtime, by the application in different ways. The first and most obvious option is to directly specify the full utility function, which takes the relevant metrics as parameters. The programmatically simplest way is to provide the function as a black box, which is evaluated by the adaptive system at specific points. More efficient optimization may be achievable by providing the utility functions in an analytic closed form with certain properties, such as continuous differentiability or convexity. This, however, requires a representation of the analytical form that can be evaluated by the optimizer, which causes a higher specification and implementation overhead than a black-box function. Additionally, the system model that stands between the utility function and the actual low-level optimization operation (cf. Section 6.1) will in many cases destroy the utility function's analytic properties (e.g., convexity [26, Section 1.1.2]) so that the optimizer can no longer benefit from them. Finally, if the application can freely update the utility functions during runtime, a distributed optimization approach may require exchanging the respective utility functions among nodes to estimate each other's utility. This introduces additional communication overhead.

Besides setting and updating the whole utility function, a second option is to use additional parameters to the utility function, which reflect application or system state that influences the overall utility. An example for this is whether the application is being actively used in the foreground at the moment or just idling in the background and waiting for incoming notifications. In such case, an 'active' parameter could modify the utility function to set lower requirements to the notification delay without the need for replacing the utility function. Additionally, such parameter can be set by the system to distinguish different objects to which the utility relates. Where necessary for the distributed optimization algorithm, those parameters, typically being scalar values, can be exchanged among nodes with relatively little overhead.

For InterestCast, we use a combination of the two described approaches. The utility function can be provided by the application as a black box function. It is assumed, however, to be constant during runtime and that all nodes use the identical function. This allows nodes to predict utility changes for a given node without the need for exchanging their respective utility functions. Instead, adaptation of utility is achieved by introducing an additional parameter to the latency utility function, the *interest level*. This interest level determines the urgency of the delivery of events from a particular source. We therefore use the utility function $U_{\text{lat}}(\mathscr{L}, \iota)$, with $\mathscr{L}$ being the event delivery latency and ι the respective interest level. For more details on the modeling of InterestCast's utility functions, refer to Section 5.3.

4.3 Topology and Routing

InterestCast's purpose is to provide many-to-many event dissemination among participants in the network. As an application-level middleware, it builds an overlay topology on top of an existing layer 3 (IP) network. The goal is to optimize event routing according to application needs and network conditions. In its basic function, InterestCast is supposed to work in a peer-to-peer fashion

without any dedicated infrastructure. It is therefore necessary to distribute the required functionality, most importantly event multiplication and filtering as well as the optimization of this process, among the peers in a scalable way.

Approaches assuming dedicated servers for their operation usually have both the filtering and the multiplication on the servers, delivering the clients only their individual set of events. Typical examples are broker-based publish/subscribe systems (cf. Section 3.4), where the brokers manage the subscription filters and send notifications for all their events individually. In application-layer multicast solutions (cf. Section 3.2) using dedicated multicast service nodes, those are responsible for filtering based on multicast group membership and message delivery. All such centralized approaches rely on server instances that need to have enough capacity to serve the desired number of users.

With the simple direct delivery schemes used in many peer-to-peer overlays for virtual environments (cf. Section 3.6), the source node is responsible for filtering by managing the interest set (i.e., receiver list) and multiplication by sending the event message to each receiver individually. This works well and with low latencies as long as the respective sending node has enough uplink capacity to dispatch its messages to all receivers within reasonable latency bounds. When a single event is to be disseminated, the messages to the individual receivers instantaneously queue up. The higher the number of receivers and the lower the sender capacity, the longer will it take to dispatch all messages. To mitigate this problem, peer-to-peer multicast solutions distribute the message multiplication, so that nodes with higher capacity or less load can take over part of the work. In most cases, this is achieved by building a multicast tree. As already discussed in Section 3.2, there are many options for how the tree is built with different trade-offs with respect to group size, group and network dynamics, bandwidth and latency constraints, etc.

First of all, the tree can either be considered on its own or together with the trees from other sources. Many application-layer multicast solutions assume a limited number of groups of which each has a tree, and therefore only consider a single tree. InterestCast, however, assumes each node to be a distinct source, so that the interaction of dissemination trees, particularly concerning shared edges, must be taken into account. Edge sharing is relevant because it allows aggregating packets thereby reducing packet overhead.

The construction, maintenance, and optimization of the tree can be performed either centrally at the source or in a distributed way by the nodes in the tree. The main advantage of the source-based construction is that it can be done efficiently using a central algorithm. The tree topology information then has to be transmitted along with the data along the tree. This allows highly dynamic trees as re-configurations are just propagated with the actual data. The encoding of the whole topology, however, can cause significant overhead, especially when the payload is small and the tree becomes large. More importantly, the central tree construction requires that all necessary information is available at the source. While being informed about all receivers and their basic properties can be reasonable, latency optimizations require knowledge about network latencies between pairs of receivers, which is $O(n^2)$ state that needs to be updated regularly. Furthermore, potential forwarders that are not in the receiver set might not be known to the source.

The alternative is to let the nodes in the tree optimize the routing based on their local knowledge. This way, each node can choose to rearrange its position in the dissemination structure in certain

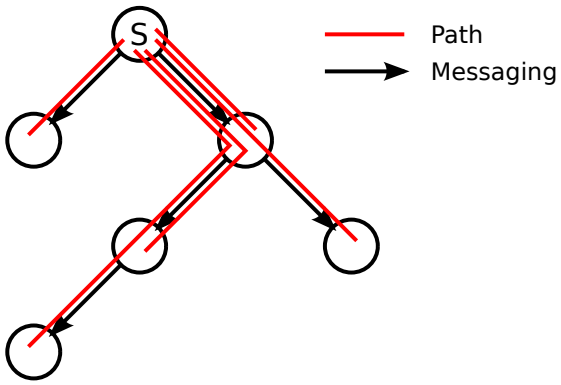


Figure 4.4.: Virtually, InterestCast treats all paths from source (S) to the individual destinations separately (red), while actual messages are only transmitted on the black arrows.

ways. In a tree structure, this could be moving up or down the tree, i.e., towards or away from the source, or switching siblings like in BTP [77] (cf. Section 3.2). Doing so requires that the node performing a local switching operation has the knowledge necessary for projecting the effects of the change and the resulting performance impacts on the affected nodes. The easiest way is to let the nodes act selfishly without considering impact on others, but optimizing the overall system for certain requirements is only achievable under certain conditions.[4] Forwarding nodes should therefore be informed about the requirements and the fulfillment of those and take these into consideration for their local decisions.

On the topology and routing level, InterestCast therefore treats each path from source to destination as an individual unicast stream, as illustrated in Figure 4.4. This allows measuring the end-to-end delay and optimizing each individually, even though this introduces some extra overhead. On the plus side, application-level unicast messages and messages to subgroups can use the same routing mechanism as the regular dissemination. The model and implementation details of InterestCast's routing can be found in Sections 5.1 and 7.3, respectively.

4.3.1 Topology Construction

Selecting appropriate forwarding configurations requires knowledge about the possible forwarders. Considering all nodes in the network is too expensive in most cases, particularly because of the aforementioned $O(n^2)$ inter-node latency state. Some application-layer multicast systems, such as Narada [41] (cf. Section 3.2), use a reduced mesh overlay that is managed by an intermediate layer. This mechanism limits the number of neighbors to a certain degree and therefore reduces the neighbor state that has to be exchanged. Narada uses network latency as the criterion for mesh neighbor selection. This way, the mesh can be fit to the underlay topology and therefore

[4] In game theory, *potential games* [118] have the property that if each participant acts (rationally) selfishly, the global optimum can be reached. This property, however, is in general not given in our case. One of the reasons is that nodes manage flows of which they are neither source nor destination. Such flow would be of no interest for a selfishly acting node and it would certainly just drop it to save costs. We therefore do not consider nodes acting selfishly, but rather to be collaborative with a common goal.

reduces the forwarding options to the well-suited candidates. On the other hand, it introduces an additional management layer, and changes in the mesh topology must be passed up to be reflected in the dissemination tree. Furthermore, the mesh topology potentially constrains the options for the most direct routes.

Assuming very dynamic group membership and network conditions, we take a different approach. InterestCast uses the neighbors with mutual interest as potential forwarding candidates. This approach is promising with a high clustering of interest in the interest graph (cf. Section 5.1), which is the case in virtual reality scenarios [101] (see also Section 9.3.1). Therefore, with nodes knowing about their neighbors of interest as well as interested neighbors, those are likely to have a high ratio of common interests. Common interests allow piggybacking update messages to each other, making those neighbors well-suited forwarding candidates.

4.4 Optimization Strategy

Based on the utility functions discussed above, InterestCast optimizes its message routing configuration. Analogously to the tree construction, this optimization can be done either centrally, e.g., by the source node, or decentralized among all participating nodes. Consequently, our choice is to allow all nodes on a path from source to destination (except for the destination node itself) to adapt the path to increase the overall utility based on their local knowledge. This unburdens the source node from the need for maintaining up-to-date state of all affected nodes.

As introduced above (Section 4.2), InterestCast uses two utility functions. First, there is the utility for the end-to-end delivery latency that is applied to each individual path. Secondly, the cost[5] for bandwidth demand allows for a penalization of highly loaded nodes (with respect to their link capacity). For the exact definition of the optimization objective, refer to Section 5.5.

Unlike heuristic tree building approaches such as Hyphen [3] (cf. Section 3.5), InterestCast starts with an initial configuration that only uses direct routes from source to destination. The idea behind this concept is that newly initiated connections are set up quickly without the need for first being inserted into a dissemination tree, therefore providing best-effort delivery under dynamic conditions. Subsequently, the routing is incrementally updated by inserting intermediate forwarders wherever this increases the overall utility. The detailed description of the respective operations can be found in Section 6.1.

The abstract distributed optimization algorithm used in this process is sketched in Algorithm 1. Each node regularly invokes the OptimizeStep function, which selects at most one operation to change the routing configuration. Since each node does this individually based on local knowledge, it is a local greedy optimization approach.

In its basic version, each iteration of OptimizeStep first enumerates the possible configuration changes (transitions). Each option is then evaluated with respect to the expected change in the system (PredictState) and the consequent change in the system behavior (PredictMetrics) and utility (Utility). Since the global state of the system is not known to the local node, the system

[5] Recall that we do not strictly distinguish between utility and cost, assuming that one can be represented as the negation of the other.

Algorithm 1 The basic local greedy optimization algorithm used by InterestCast. In each iteration the function OptimizeStep is invoked, which performs one routing change, if this is expected to improve the utility.

```
1: A ← current system state
2: m_A ← measurements from current state
3: u⁻ ← transition cost/threshold
4: procedure OptimizeStep
5:     𝒯 ← EnumeratePossibleTransitions(A)        ▷ enumerate all possible local transitions
6:     T* ← ∅                                     ▷ best transition (initialize with none)
7:     Δu* ← 0                                    ▷ projected utility difference of best transition
8:     for T ∈ 𝒯 do
9:         B ← PredictState(A, T)                 ▷ predict the system state after transition T
10:        m_B ← PredictMetrics(m_A, B)           ▷ predict metrics changes after transition T
11:        Δu ← Utility(m_B) − Utility(m_A)       ▷ utility difference to current state
12:        if Δu > Δu* then                       ▷ if this is the best option so far, select it
13:            T* ← T
14:            Δu* ← Δu
15:    if Δu* − u⁻ > 0 then                       ▷ perform best transition, if any
16:        ExecuteTransition(A, T*)
```

behavior (m_B) after the transition is estimated based on the current behavior (m_A).[6] The predicted utility change Δu is used to quantify the value of the respective option. The transition cost parameter u^- can be used to set the minimum change in utility that is required to perform the transition. Finally, the algorithm selects the transition that has the highest expected change in utility, if there is at least one that is above zero.

4.4.1 System Performance Model

A prerequisite for the presented approach is the ability to predict the influence of the system reconfiguration on the system behavior, i.e., on the performance metrics, with reasonable precision. This implies that there is the need for a good system model to predict its performance. The utility prediction process from Algorithm 1 is illustrated in Figure 4.5.

For InterestCast, the metrics to be predicted by the system model are end-to-end latency and bandwidth demand. The main input factors of the system model are the nodes' link capacities, their usage, and network latencies between the nodes. As discussed in Section 4.2, network latencies can only be measured by the nodes and are assumed to be load independent, though they may vary over time. Link capacities are assumed to be known by the respective node, usually constant, but possibly also varying over time.

The model for bandwidth demand is rather simplistic. We use the relative bandwidth demand, which is computed as the ratio of bandwidth demand and link capacity. Predicting the usage of

[6] We abstract the system behavior as a vector of performance metrics m_S that is assumed to be measurable when the system is in state S. Refer to Section 6.2 for more details.

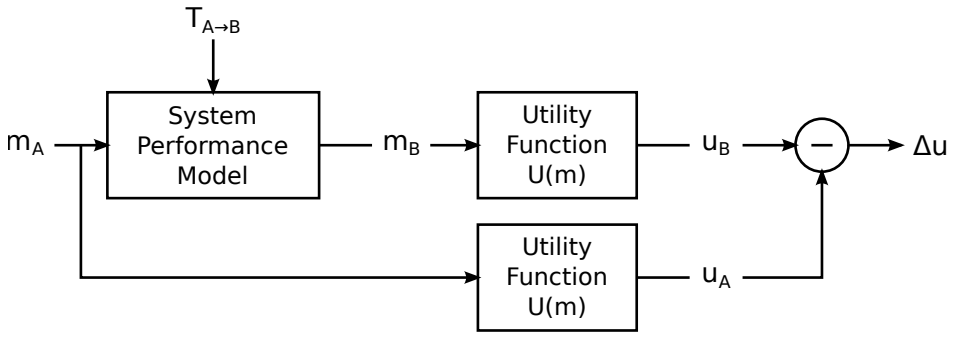


Figure 4.5.: The basic prediction of utility changes caused by a transition $T_{A\to B}$ from current state A to a potential state B. m denotes the system performance metrics in a given state, $U(m)$ is the corresponding utility value.

routing reconfigurations requires the averaged throughput of the respective path. Therefore, the throughput of all paths needs to be monitored. The exact effect of message aggregation is hard to predict because it depends on message content and timing. We therefore add a penalty for additional links (i.e., a hop between two nodes that has not been used before) and inversely a reward for the removal of a link. The intuition behind this is that any existing link can potentially be used for aggregation.

End-to-end latency is modeled as a composition of network latencies between pairs of nodes on each hop and the queuing delay on the nodes before each hop. The queuing delay is calculated using standard queue models and depends on the corresponding node's link utilization. Hence, the relative bandwidth demand also affects the projected end-to-end latency and is considered as a factor even if it is not directly associated with a cost by the application (cf. Section 4.2). The concrete performance model for end-to-end latency can be found in Section 5.4.

4.4.2 Optimization Objective

In contrast to a global optimization algorithm, the local optimization approach restricts the possible objectives to sums of utilities. Intuitively, this is due to the incremental steps towards any utility improvement. Hence, the maximization of minimal utilities or other fairness-related aspects cannot be covered. For a best-effort system, however, this appears to be a reasonable choice, because the non-fulfillment of the utility of one participant should not penalize the utility of another.

Furthermore, since InterestCast allows for specifying utility functions for two aspects, that is delivery latency and relative bandwidth demand, we have to take into account how to deal with multiple objectives. One option is to consider them separately, prioritizing one. This is done, for instance, by Narada [41] (cf. Section 3.2). Only if more than one alternative exists with the highest value for the first factor, the second factor is considered. Depending on the concrete use case, this may barely be the case. Another option is to treat the overall utility as a function of all factors, e.g., as a weighted sum. The weights allow trading different factors against each other, treating them as linearly comparable. For InterestCast, we choose this approach. Delivery latency and relative bandwidth demand are each included in the sum with a weight, see Section 5.5.

5 System Model and Problem Formalization

In this chapter, we introduce and explain the fundamental system model and discuss our considerations and assumptions. Afterwards, we define the formalized problem and the overall optimization problem.

5.1 Basic System Model

Definition 1. V denotes the set of nodes in the system.

Each node $v \in V$ represents one participant in the system. We assume that each participant (also referred to as user or player) uses an individual host in the system. The terms node, host, participant, user, and player are therefore used interchangeably in this model.

Pairs of nodes communicate by sending messages over the underlying network.

Definition 2. $\ell_{\text{net}}(v,u)$ defines the underlay network latency for messages being sent from node v to u.

The underlay network latency is usually assumed to be symmetric, i.e., $\ell_{\text{net}}(v,u) = \ell_{\text{net}}(u,v)$. For the algorithms presented in this thesis, this is generally not necessary. In a practical implementation, however, only the round-trip time (RTT) can be measured, so that the symmetry assumption has to be made.

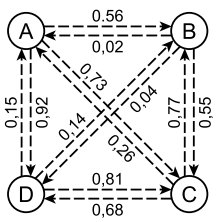
Definition 3. $\mathcal{B}_v$ defines the capacity, i.e., maximum bandwidth, of node v's uplink.

To keep the model simple, we only consider uplink capacity. Most consumer and residential internet connections are asymmetric with much larger downlink than uplink capacities [52], so that the downlink is rarely the bottleneck. Furthermore, we assume that the relevant bandwidth bottleneck is close to the user rather than on the core network.

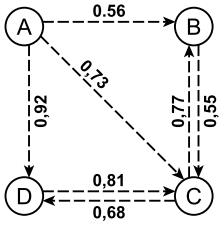
Definition 4. $b_v := \frac{\text{bandwidth demand of } v}{\mathcal{B}_v} \geq 0$ denotes the relative uplink bandwidth demand of node v.

The term *bandwidth demand* refers to the bandwidth that would be used to send all data in the current configuration and with the given workload without loss of any event. Unlike the effective *uplink utilization*, the relative bandwidth demand b_v can exceed the value of 1. Therefore, if $b_v > 1$, v cannot send fast enough and has to drop events. This obviously should be avoided. Values close to one usually imply heavy queuing and thus are undesired as well.

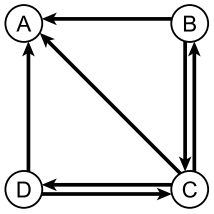
Each node generates events with updates from its own state. This could be a position update in a virtual world or a status notification. Other nodes may be more or less interested in the updates from a particular node:



(a) The graph induced through the interest function applied on all pairs of nodes.



(b) Applying the threshold of $\tau = 0.5$ determines the subscription graph G.



(c) The resulting notification or event flow graph.

Figure 5.1.: Example of a subscription graph being generated from a given interest function I

Definition 5. The *interest function* $I : V \times V \to [0,1]$ quantifies the interest level of one node in another. $I(v,u)$ is the interest of node v in the updates of node u. Generally, the interest level is continuous, but in simple cases, it can be binary. Then, the interest function is defined as $I : V \times V \to \{0,1\}$.

Figure 5.1a shows an example of the interest between nodes.

Definition 6. The *interest set* $\mathscr{I}_v$ of a node $v \in V$ is defined as

$$\mathscr{I}_v := \{u \in V \mid I(v,u) \geq \tau\},$$

where $\tau \in (0,1]$ is the interest threshold.

τ is thus the minimum interest level required for receiving any updates from the corresponding node. For each node $u \in \mathscr{I}_v$, v is subscribed for the updates of u. Interpreting $\mathscr{I}_v$ as adjacency lists, we can define the subscription graph:

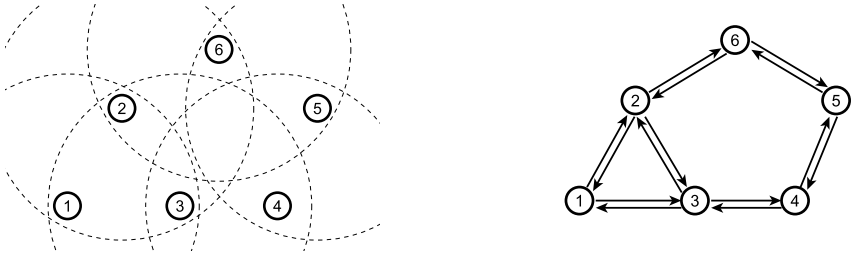
Definition 7. $G = (V,E)$ is the directed *interest* or *subscription graph*, where

$$E := \{(v,u) \in V \times V \mid I(v,u) \geq \tau\}.$$

$\tau \in (0,1]$ is the interest threshold, i.e, the minimum interest level required for receiving updates from the corresponding node. Therefore, v is subscribed to u's updates iff $(v,u) \in E$ (Figure 5.1b). Although subscriptions are binary, i.e., a node is either subscribed to another or is not, the continuous level of interest serves as an indicator for the delivery priority.

For each subscription (v,u), there will be a corresponding *event flow* from u to v (Figure 5.1c). Events may be sent directly from u to v or via any number of intermediate *forwarder* nodes. In this model, we assume that events are sent on a regular basis and do not exceed a predefined maximum throughput per flow. The sequence of (forwarding) steps that messages of a particular flow take is defined as their *path*.

Figure 5.2 shows an example of a virtual world where the players have a circular AOI. This model can be extended in that the continuous interest level is determined by factors like distance, focus, and interaction [21]. Thus, the updates of distant, hidden, or by other means less important players are less critical. It also becomes apparent that the interest levels are highly dynamic over time. A sudden interaction, for example, may instantly raise the interest in a particular player.



(a) Equally-sized AOI circles for all nodes. **(b)** The resulting subscription graph.

Figure 5.2.: An example of how a subscription graph is generated from positions in a 2-D virtual world

5.1.1 Event Delivery Paths

Events can be routed via one or more intermediate nodes between their source and destination.

Definition 8. The ordered set of nodes $\mathscr{P}_{v\to u} := (v_0 = v, v_1, \ldots, v_n = u)$ defines the *path* of length $n \in [1, |V|-1]$ of events delivered from v to u.

The concrete instances of $\mathscr{P}$ depend on the nodes' routing configurations, but are the same for all events with the same source and destination.[1] The set of events being routed along the same path is called a *flow*.

The end-to-end delay of a given event is the sum of network transmission delays on each hop (ℓ_{net}), and the queuing delay on the node before each hop (ℓ_{q}). The processing time on the nodes is not modeled as a separate parameter because we assume it to be insignificant in comparison with the other two components. Thus the end-to-end latency of a single n-hop event delivery is calculated as

$$\mathscr{L}_{v\to u} = \sum_{i=0}^{n-1} \big(\ell_{\text{q}}(v_i) + \ell_{\text{net}}(v_i, v_{i+1})\big). \tag{5.1}$$

5.2 Event Classes

We distinguish two classes of events: *update events*, which carry state updates, and *instant events*, which are triggered by actions at specific points in time. The two classes have slightly different requirements and utilities.

5.2.1 Update Events

Update events serve the purpose of synchronizing state among entities. In our model, each entity manages mutable state which is of potential interest for other entities.

[1] To keep things simple, we do not distinguish different event types on potentially different paths. This could be added to the model as additional instances of the optimization problem.

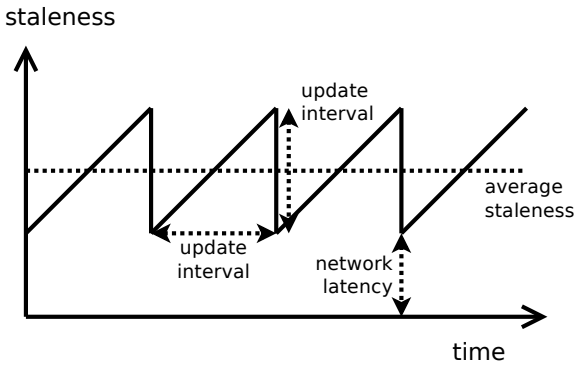


Figure 5.3.: Illustration of staleness over time with periodic updates

The goal is to keep the information about the changing state of one entity as accurately as possible at all interested entities. We use the metric *staleness* (the inverse of *freshness*) to quantify the fulfillment of this goal. Staleness is the age of state available at a given node, assuming constantly changing state. We assume that each network node manages exactly one entity and each entity manages a single state variable. From the state S_v of node v, an interested node u keeps a copy of the state, S_v^u. The staleness of that state at time t is defined as

$$\mathscr{S}_{S_v^u}(t) := \min\left\{\Delta t \mid S_v^u(t) = S_v(t-\Delta t)\right\},$$

i.e., the time difference Δt between the time t and the last time for which the state available at node u was current.

Staleness is influenced by two factors: update rate and update dissemination latency. With a constant update rate the staleness over time has a sawtooth shape, as illustrated in Figure 5.3. Intuitively, the average staleness can be calculated as

$$\overline{\mathscr{S}} = \overline{\mathscr{L}} + \frac{\overline{T}}{2}, \tag{5.2}$$

where $\overline{T}$ is the average update interval (i.e., $\frac{1}{\overline{T}}$ is the update rate) and $\overline{\mathscr{L}}$ is the average event delivery latency. Hence, update events have two parameters that can be used to tune staleness. Moreover, this model incorporates the effects of update message losses: $\overline{T}$ just has to be defined as the effective update inter-arrival time at the destination.

Update events may contain the full state or the delta-encoded difference. For full update messages that are sent regularly, a message loss only transiently increases staleness. In such case, messages can be transmitted using unreliable protocols and can be purposefully omitted in overload situations. With small state data, such as positions, delta encoding is usually of little benefit. In the following, we therefore assume full state updates and with it the option to skip state update events where necessary. This increases the staleness of the state at the receiver, which is counted as a penalty in the staleness metric.

Some aspects of entity state can be predicted by other entities to a certain degree, in the simplest case by just extrapolating the trajectory of a value. An example for this is position state. A common technique for synchronizing player positions is *dead reckoning* [129], which uses inter-

and/or extrapolation of positions of neighboring players and omits position updates as long as the extrapolations remain within certain bounds. Applying such a technique reduces the meaningfulness of the staleness metric. For those cases, either a position error metric [69] can be used or just the event delivery like for instant events (see below).

5.2.2 Instant Events

Instant events are triggered upon actions at a specific point in time that are of relevance for other entities. Those events can, but do not have to be coupled to a state change. Such events are used for direct interaction in a virtual environment, like shooting in a multiplayer game or control commands that need to be disseminated, for example. Since actions happen at one instant of time, they should be delivered to the receivers as soon as possible. The metric to quantify this is the delivery latency $\mathscr{L}$.

5.3 Application Utility Functions

The primary objective from the application perspective is to optimize event dissemination latency based on the application-specific end-to-end latency cost, i.e., the utility for meeting a certain deadline. The major constraints are network latencies between the nodes (ℓ_{net}) and the nodes' uplink capacities ($\mathscr{B}$). In addition, we consider the cost of network usage in terms of bandwidth demand. This can be used, for instance, where energy savings from reducing transmissions are desired or spare capacity is to be used by other applications. Although considered a cost, the network usage is also incorporated as a utility function to gain a consistent model.

The application objectives are handed over to the system as utility functions.

Definition 9. The application defines its utility function for event delivery latency as $\mathscr{U}_{\text{lat}} : \mathbb{R}_0^+ \times [0,1] \to \mathbb{R}$.

$\mathscr{U}_{\text{lat}}(\mathscr{L}, \iota)$ returns the utility of events of interest ι being delivered with latency $\mathscr{L}$. For most applications, this function is monotonically decreasing, since lower latencies are usually considered better. Furthermore, high interest values are considered to imply a stronger negative impact on utility than low interest values. Concrete examples are given in Section 6.5.

For update events (Section 5.2.1), the application can alternatively specify a staleness-based utility function $\mathscr{U}_{\text{sta}}(\mathscr{S}, \iota)$, equivalent to the latency utility.

Definition 10. The application's utility function for uplink bandwidth demand is defined as $\mathscr{U}_{\text{bw}} : [0,1] \to \mathbb{R}$.

$\mathscr{U}_{\text{bw}}(b)$ returns the utility of the relative uplink demand b (Definition 4). For nodes with different individual demand utilities, $\mathscr{U}_{\text{bw}}$ can be differentiated by node. We omit this in the following for the purpose of simplicity, although it can be added to the model with little effort. Concrete examples of bandwidth demand utility functions can be found in Section 6.5.

Even though the utility functions map to $\mathbb{R}$, we only require utilities to be ordinal. That is, for any two utility values, we can decide whether one is better (i.e., greater) than, worse (i.e., less)

than, or equal to the other. In contrast to a cardinal utility, however, a doubled utility value does not necessarily mean 'twice as good'. This relaxation simplifies the definition of utility functions for the developer.

Note that the image of the utility functions is defined in $\mathbb{R}$ with no constraints. For a reasonable trade-off between the two, i.e., latency and bandwidth demand, it is, however, up to the application developer to define functions with comparable images. This can be achieved by normalizing the utility functions with respect to certain reference values, such as a target latency or critical points of the relative demand. Refer to Section 6.5 for more details.

5.4 Performance Model

The application utility functions introduced in the previous section are supposed to give the application developer the option to choose between trade-offs in abstracted and declarative fashion. Ideally, this should be done without the need for knowing about the effects introduced by the underlying network, such as interdependencies between link utilization and queuing delays. Therefore, the system model should incorporate those effects where possible, so that they do not need to be considered in the utility functions. This section provides the models for two effects considered important for our system, namely queuing delay and update event loss.

5.4.1 Queuing Delay

As defined in Equation 5.1, the forwarding latency on each hop consists of the local queuing delay ℓ_q and the underlay network latency ℓ_net. While ℓ_net is given as a system parameter, which is in the real implementation measured by the system, ℓ_q depends on the sending node's link utilization. We use a queuing model to make estimates on the queuing delay.

Modeling the outgoing event scheduler on each node as an M/D/1 queue,[2] the average queuing time at a given node v being the sender for the respective hop is calculated as (cf. [24, pp. 245, 256])

$$\ell_\mathrm{q}(v) = \overline{T}_v = \overline{W}_v + \frac{1}{\mu_v} \tag{5.3}$$

$$= \frac{\overline{Q}_v}{\lambda_v} + \frac{1}{\mu_v} \tag{5.4}$$

$$= \frac{\rho_v^2}{2\lambda_v(1-\rho_v)} + \frac{1}{\mu_v} \tag{5.5}$$

$$= \frac{\rho_v}{2\mu_v(1-\rho_v)} + \frac{1}{\mu_v} \tag{5.6}$$

$$= \frac{2-\rho_v}{2\mu_v(1-\rho_v)}, \tag{5.7}$$

[2] We assume an exponential arrival distribution and a constant service time, since the packet output is limited by a constant rate limiter and packets are of similar size. This way, we are somewhat too optimistic with respect to queue length, since the general queue length $\overline{Q} = \frac{\rho^2}{1-\rho} \cdot \frac{1+c_B^2}{2}$ [24, p. 256, Eq. (6.51)] increases with the service time variance coefficient c_B.

where $\overline{T}$ is the average time in the queuing system, $\overline{W}$ is the average waiting time in the queue, $\overline{Q}$ is the average queue length, $\rho_v = \frac{\lambda_v}{\mu_v}$ is the uplink utilization of node v, and λ_v and μ_v are the rate limiter's effective packet input and output rates, respectively. ρ_v can be measured at the rate limiter. μ_v is calculated as the current byte rate setting at the rate limiter divided by the average packet size.

The overall utility function with respect to latency therefore also depends on the link capacities and utilizations of all participating nodes. Given the application-level event latency utility function $\mathcal{U}_{\text{lat}}$, the effective event latency utility estimation evaluates as

$$U^*_{\text{lat}}(\mathcal{P}, \iota) := \mathcal{U}_{\text{lat}}\left(\sum_{(v,u)\subseteq\mathcal{P}} \ell_{\text{q}}(v) + \ell_{\text{net}}(v,u), \iota\right). \tag{5.8}$$

5.4.2 Update Event Loss

In cases where the data volume that needs to be transmitted by a node exceeds its uplink capacity, dropping messages is inevitable. The loss of update events, which we consider here, will lead to an increase of staleness. Using a staleness-based utility function for update events, the expected increase of staleness due to update loss can be incorporated.

With b_v being the relative uplink bandwidth demand of node v (i.e., possibly $b_v > 1$), we approximate the loss probability of a single message outgoing from v with

$$P_{\text{loss}}(v) = \max\left\{1 - \frac{1}{b_v}, 0\right\}. \tag{5.9}$$

Assuming independent loss probabilities on each hop along path $\mathcal{P}$, we obtain the loss probability on the full path as

$$P_{\text{loss}}(\mathcal{P}) = 1 - \prod_{i=0}^{|\mathcal{P}|-1} (1 - P_{\text{loss}}(\mathcal{P}(i))) \tag{5.10}$$

$$= 1 - \prod_{i=0}^{|\mathcal{P}|-1} \min\left\{\frac{1}{b_{\mathcal{P}(i)}}, 1\right\} \tag{5.11}$$

The effective update inter-arrival time (cf. Section 5.2.1) can therefore be approximated with

$$\hat{T}(P_{\text{loss}}) = T \cdot \frac{1}{1 - P_{\text{loss}}} \tag{5.12}$$

and the expected staleness (cf. Equation 5.2) is modified as

$$\mathcal{S}_{v\to u} = \mathcal{L}_{v\to u} + \frac{\hat{T}(P_{\text{loss}}(\mathcal{P}_{v\to u}))}{2} \tag{5.13}$$

$$= \mathcal{L}_{v\to u} + \frac{T}{2 \cdot \prod_{v'\in\mathcal{P}_{v\to u}\setminus u} \min\left\{\frac{1}{b_{v'}}, 1\right\}} \tag{5.14}$$

$$= \mathcal{L}_{v\to u} + \frac{T}{2} \prod_{v'\in\mathcal{P}_{v\to u}\setminus u} \max\{b_{v'}, 1\}. \tag{5.15}$$

We can therefore modify the staleness utility function such that

$$U^*_{\text{sta}}(\mathscr{P}_{v\to u}, \iota) := \mathscr{U}_{\text{sta}}\left(\mathscr{L}_{v\to u} + \frac{T}{2}\prod_{v'\in\mathscr{P}_{v\to u}\setminus u}\max\{b_{v'},1\}, \iota\right). \tag{5.16}$$

Combining this with the queuing model from above (Equation 5.8), we finally obtain

$$U_{\text{sta}}(\mathscr{P}_{v\to u}, \iota) := \mathscr{U}_{\text{sta}}\left(\frac{T}{2}\prod_{v'\in\mathscr{P}_{v\to u}\setminus u}\max\{b_{v'},1\} + \sum_{(v,u)\subseteq\mathscr{P}}\ell_{\text{q}}(v) + \ell_{\text{net}}(v,u), \iota\right). \tag{5.17}$$

5.5 Optimization Problem

Finally, to define a single objective function for the optimization, it is necessary to combine the utility functions for latency and bandwidth demand.

The factors w_{lat} and w_{bw} specify the weights of latency and bandwidth demand utility respectively. They can be used to adjust the trade-off between the two. The total utility function (latency-based) is consequently defined as

$$U(\mathscr{P}) := w_{\text{lat}} \cdot \sum_{(v,u)\in E} U_{\text{lat}}(\mathscr{P}_{u\to v}, I(v,u)) \; + \; w_{\text{bw}} \cdot \sum_{v\in V} U_{\text{bw}}(b_v) \tag{5.18}$$

and the staleness-based version as

$$U(\mathscr{P}) := w_{\text{lat}} \cdot \sum_{(v,u)\in E} U_{\text{sta}}(\mathscr{P}_{u\to v}, I(v,u)) \; + \; w_{\text{bw}} \cdot \sum_{v\in V} U_{\text{bw}}(b_v). \tag{5.19}$$

Summing up the utilities of the nodes' link utilizations and the latencies this way has the effect that a changing ratio of interest graph edges and nodes ($\frac{|E|}{|V|}$) will influence the weight ratio of the two components. The ratio depends on the density of the interest graph and can therefore fluctuate during runtime. An option would be to normalize the two components by using the factors $\frac{w_{\text{lat}}}{|E|}$ and $\frac{w_{\text{bw}}}{|V|}$ instead of w_{lat} and w_{bw}, respectively. This would, however, decrease the weight of a single edge's staleness utility with the total number of edges. While this could be a reasonable decision for some applications, we do not want to make such assumption here. Instead, the application can reduce the interest levels from $I(v,u)$ accordingly to achieve the same effect, if desired.

Based on the system model and utility functions described above, we can now formulate the formal optimization problems considered in this thesis:

$$\max \quad w_{\text{lat}} \cdot \sum_{(v,u)\in E} U_{\text{lat}}(\mathscr{P}_{u\to v}, I(v,u)) \; + \; w_{\text{bw}} \cdot \sum_{v\in V} U_{\text{bw}}(b_v), \tag{5.20}$$

based on dissemination latencies, and the staleness-based version

$$\max \quad w_{\text{lat}} \cdot \sum_{(v,u)\in E} U_{\text{sta}}(\mathscr{P}_{u\to v}, I(v,u)) \; + \; w_{\text{bw}} \cdot \sum_{v\in V} U_{\text{bw}}(b_v). \tag{5.21}$$

In words, we maximize the sum of utilities of the end-to-end latencies and stalenesses, respectively, over all edges in the interest graph and the utilities of the nodes' bandwidth demands. The latency and staleness utility functions are further dependent on the interest level on the individual interest graph edge.

5.6 Global Solutions Using Integer Programming

To gain insights about the best possible solutions for given static constellations, we first aim for global solutions to the problem formulation described above (Section 5.5). These serve as reference solutions (benchmarks) for the results of the local optimization processes. Global optima, being the best possible solutions, allow us to determine how close to the optimum we get using the InterestCast's local algorithm.

To efficiently find global optima for given static constellations, we use standard solver software. For this, we transform the problem into an integer programming (IP) problem [123], more specifically into an integer linear program (ILP) and a mixed integer nonlinear program (MINLP, [27]). The ILP program serves as a reference that is easier to solve due to its linearity, while the MINLP program considers non-linear utility functions.

To transform the problem into a linear program, we have to make certain simplifications. First of all, we only consider static scenarios, i.e., a fixed interest graph, fixed event throughputs, fixed inter-node latencies, and fixed node capacities. Further, we ignore all kinds of protocol and monitoring overhead that is necessary in a real implementation. We only consider path latencies rather than dealing with staleness influenced by varying update rates and packet loss. With a globally optimized solution, we can assume an ideal scheduling without packet drops. The event bandwidth model is simplified so that each event flow (for each $(v,u) \in E$, Definition 7) has a cost of one bandwidth unit and each overlay edge between two nodes that transports at least events from one flow costs one additional unit. This model arises from the simplified assumption that packet headers have roughly the same size as event payloads and that events from multiple streams can be perfectly combined into one packet so that they share packet headers (cf. Section 2.1). Furthermore, events from one source are assumed identical for all destinations and therefore only count as one when transmitted between two nodes.

We use the Zimpl [92] language to encode the problems programmatically and the SCIP solver [64, 1] to find the solutions.

5.6.1 ILP Problem

We start with defining the simpler integer linear programming (ILP) problem. Its solution space is defined as

$$y_{i,j,k,l} = \begin{cases} 1 & \text{if events for flow from source } k \text{ to destination } l \text{ are forwarded from } i \text{ to } j, \\ 0 & \text{otherwise.} \end{cases} \tag{5.22}$$

Therefore, for a given path $\mathscr{P}_{k\to l}$ (where $(k,l) \in E$), for all pairs of nodes on the path, corresponding entry must be set to one, i.e.,

$$y_{i,j,k,l} = \begin{cases} 1 & \text{if } (i,j) \subseteq \mathscr{P}_{k\to l}, \\ 0 & \text{otherwise.} \end{cases} \tag{5.23}$$

For now, we ignore the spare capacity utility and only consider delivery latency. Node capacities will be modeled as constraints. Given the utility function based on the event delivery latency $\mathscr{L}$, $U_{\text{lat}}(\mathscr{L})$, we can define the objective function as

$$\sum_{(k,l)\in E} U_{\text{lat}}(\mathscr{L}_{k\to l}) = \sum_{(k,l)\in V\times V} U_{\text{lat}}\left(\sum_{(i,j)\in V\times V} y_{i,j,k,l} * \ell_{\text{net}}(i,j)\right). \tag{5.24}$$

This expression using an arbitrary utility function, however, is not generally solvable using an ILP solver. For this reason, we confine ourselves to linear utility functions and the special case $U(\mathscr{L}) = a \cdot \mathscr{L}$. Being a constant factor, a can be moved outside of the sum. This way, the overall utility function simplifies to

$$\sum_{(i,j,k,l)\in V\times V\times V\times V} y_{i,j,k,l} \cdot a \cdot \mathscr{L}_{i,j} = a \cdot \sum_{(i,j,k,l)\in V\times V\times V\times V} y_{i,j,k,l} \cdot \mathscr{L}_{i,j}. \tag{5.25}$$

Since we want to minimize latency, a is assumed to be negative. Furthermore, being a constant factor in the objective function, we can simply ignore its absolute value. The final objective is therefore to minimize the sum of latencies of all paths:

$$\min \sum_{(i,j,k,l)\in V\times V\times V\times V} y_{i,j,k,l} * \mathscr{L}_{i,j} \tag{5.26}$$

s.t.

$$\forall i \in V, \sum_{(j,k)\in V\times V} \max_{l\in V} y_{i,j,k,l} + \sum_{j\in V} \max_{(k,l)\in V\times V} y_{i,j,k,l} \le \mathscr{B}_i$$

Nodes do not exceed their link capacities.[3]

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \forall i \in V \setminus \{k,l\}, \sum_{j\in V} y_{j,i,k,l} = \sum_{j\in V} y_{i,j,k,l}$$

In-degree = out-degree, except for source and destination.

$$\forall i \in V, \sum_{(k,l)\in V\times V} y_{i,i,k,l} = 0$$

Nodes do not route to themselves.

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{j\in V} y_{k,j,k,l} = 1$$

Source has exactly one outgoing flow.

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{j\in V} y_{j,k,k,l} = 0$$

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{i\in V} y_{i,l,k,l} = 1$$

Dest. has exactly one incoming flow.

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{i\in V} y_{l,i,k,l} = 0$$

The Zimpl [92] script of this problem formulation can be found in Appendix A. Since maximum and minimum operations as used in the above description cannot be expressed directly in this form, the script introduces the auxiliary variables x and z, which are constrained with respect to y.

[3] As described above, we count one bandwidth unit per overlay edge used ($\max_{(k,l)\in V\times V} y_{i,j,k,l}$) plus one unit per outgoing flow of an individual source ($\max_{l\in V} y_{i,j,k,l}$).

5.6.2 MINLP Problem

To overcome the limitation to linear objective functions and constraints of the ILP formulation, we extend it to a mixed integer nonlinear program (MINLP). The MINLP problem allows us to use non-linear utility functions. Therefore, we can replace Equation 5.26 of the ILP problem with the original objective function (Equation 5.20). In the Zimpl code (Appendix A), we work with cost functions ($C_{\text{lat}} = -U_{\text{lat}}$, $C_{\text{bw}} = -U_{\text{bw}}$) instead of utility functions, using simple negation for the translation between utility and cost functions. We further omit the queuing model in this formulation due to limitations of the SCIP solver. A polynomial approximation of the queuing model can be provided with the link utilization cost function C_{bw} (see Appendix A).

Reusing the constraints from the ILP problem, we obtain the following optimization problem:

$$\min \quad w_{\text{lat}} \cdot \sum_{(v,u)\in E} C_{\text{lat}}(\mathscr{L}_{u\to v}, I(v,u)) \;+\; w_{\text{bw}} \cdot \sum_{v\in V} C_{\text{bw}}(b_v) \tag{5.27}$$

s.t.

$$\forall i \in V, \sum_{(j,k)\in V\times V} \max_{l\in V} y_{i,j,k,l} + \sum_{j\in V} \max_{(k,l)\in V\times V} y_{i,j,k,l} \le \mathscr{B}_i$$ Nodes do not exceed their link capacities.

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \forall i \in V \setminus \{k,l\}, \sum_{j\in V} y_{j,i,k,l} = \sum_{j\in V} y_{i,j,k,l}$$ In-degree = out-degree, except for source and destination.

$$\forall i \in V, \sum_{(k,l)\in V\times V} y_{i,i,k,l} = 0$$ Nodes do not route to themselves.

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{j\in V} y_{k,j,k,l} = 1$$ Source has exactly one outgoing flow.

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{j\in V} y_{j,k,k,l} = 0$$

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{i\in V} y_{i,l,k,l} = 1$$ Dest. has exactly one incoming flow.

$$\forall \{(k,l) \mid l \in \mathscr{I}_k\}, \sum_{i\in V} y_{l,i,k,l} = 0$$

The ILP problem solves the basic problem of minimizing the sum of latencies while keeping bandwidth demand within the capacity bounds. The MINLP problem extends this by taking the (non-linear) application-defined utility functions into account and therefore solves the Ineterest-Cast optimization problem globally.

6 Incremental Optimization Algorithm

In this chapter, we present InterestCast's incremental optimization algorithm. We further discuss the core functionalities that are necessary to run the algorithm in a distributed setting: utility estimation, neighbor information exchange, and network measurements. Finally, we discuss possible application utility functions.

6.1 Basic Algorithm

As discussed in Section 4.4, we take the approach of a local algorithm by which each node manages and incrementally optimizes its local event forwarding based on local knowledge. For that, each node maintains a forwarding table containing an entry for every outgoing event flow (cf. Section 5.1.1). Such a flow may either originate from this node, or the node is a forwarder. A forwarding table entry consists of a reference to the next hop node as well as subscription metadata like origin, destination, interest level, and the current path length and total delay.

For each new subscription, a direct path (i.e., using a direct connection between origin and subscriber) is established. All nodes continuously run iterations of their local path optimization algorithm. There are two basic operations for the optimization:

Redirect For an existing outgoing event flow, a node selects a forwarder from its local neighborhood (i.e., its interest set), that will forward the events for the next hop (Figure 6.1). This operation increases the respective path's length by one.

Shortcut If the node is neither origin nor destination, it can take itself out of the path by cutting the previous and next hop nodes short (Figure 6.2). This operation decreases the path length by one.

The optimization algorithm, as presented in [101], works incrementally. In fixed time intervals, each node repeatedly performs one iteration:

0. Let r_v be the current forwarding configuration from the point of view of the local node v, consisting of its local routes. Let

$$N_v := \{u \in V \mid (v,u) \in E \vee (u,v) \in E\}$$

be its open neighborhood in the subscription graph $G = (V,E)$ (cf. Definition 7 in Section 5.1).

(a) Initial Situation **(b)** S decides to route over U

Figure 6.1.: Redirect operation initiated by node S

(a) Initial Situation **(b)** U is no longer forwarder

Figure 6.2.: Shortcut operation initiated by node U

1. Enumerate all possible rerouting operations (redirect and shortcut). Let $\mathcal{O}$ be the set of all options:

$$\mathcal{O} := \{\text{SHORT}(s,d) \mid (s,d) \in r_v \wedge v \notin \{s,d\}\} \cup \{\text{REDIR}(s,d,u) \mid (s,d) \in r_v \wedge u \in N_v \wedge (u,s) \in E\}.$$

   $(s,d) \in V \times V$ represents a subscription of node d for s's events in v's forwarding configuration r_v.[1] $(s,d) \in r_v$ means that v is a forwarder on the current path from s to d, or $v = s$.

   Applying an operation $o \in \mathcal{O}$ will transform the current configuration r into $r' = o(r)$. The operation $\text{REDIR}(s,d,u)$ will create a redirection of the path (s,d) via u (cf. Figure 6.1). The operation $\text{SHORT}(s,d)$ will remove v from the path (s,d) (cf. Figure 6.2). Note that node v in this algorithm represents node S in the figures for the redirect case and node U in the shortcut case.

2. Find the best option according to a utility function $U : R \to \mathbb{R}$, with R being the configuration space:

$$r_{max} := \underset{r' \in \{o(r) \mid o \in \mathcal{O}\}}{\arg\max} U(r').$$

3. If $\Delta u^* = U(r_{max}) - U(r) \geq \epsilon$, then activate r_{max}, otherwise do nothing. The threshold $\epsilon \geq 0$ accounts for the transition cost. Greater values of ϵ reduce the risk of oscillations.

6.2 Utility Estimation

As motivated in Section 4.4, the utility for a potential new state (r' in Item 2 of the above algorithm sketch) cannot simply be computed, at least not without global knowledge. The PredictState function from Algorithm 1 in Section 4.4 can be trivially computed. The routing change according to the chosen transition operation (o) immediately determines the new routing configuration (r'). The critical part is the PredictMetrics equivalent, which is hidden in the utility function $U(r)$ in the above algorithm sketch.

Recalling the eventual objective function from Section 5.5, Equation 5.20, the relevant metrics in our case are the nodes' link utilizations (b_v) as well as the event delivery latencies ($\mathcal{L}_{u \to v}$) and the resulting stalenesses ($\mathcal{S}_{u \to v}$), respectively. With the global objective function being a sum of utility functions, the effect of changes in the configuration can be predicted by adding deltas (positive or negative) for those utility functions that are affected by the change. Therefore, if the overall utility (i.e., the sum of utilities) in the current system configuration is known, the prediction of the effect of a change only requires the knowledge about the metrics that are affected by the change.

[1] In a real implementation, a node's forwarding table entry stores more than only source and destination, but this is omitted here for simplicity.

6.2.1 Link Utilization

For the uplink utilization part ($\sum_{v \in V} U_{\text{bw}}(b_v)$), the differential estimation can be directly applied. For both the redirect and the shortcut operation, the uplink utilization (b_v) is changed for nodes S and U (cf. Figures 6.1 and 6.2). To predict the change in the respective utilities $U_{\text{bw}}(b_v)$, it is necessary to know the link utilizations for both nodes before and after the transition. The before state is taken from measurements. The non-local values, i.e., those for U in the redirect case and for S in the shortcut case, have to be communicated among the neighbors. The same applies to the nodes' link capacities. Finally, the throughput of the flow on the path to be changed determines the amount by which the node utilizations are shifted.

In reality, however, the exact throughput change of the individual nodes is affected by further parameters. When a node forwards multiple flows via the same neighbor, their messages may be combined (cf. Section 7.5), depending on their synchronicity and deadlines. As motivated in Section 2.1, the effective header-payload ratio, and therefore the combined gross throughput, varies greatly with the payload size. Hence, adding a flow between a pair of nodes should be considered less expensive if there is already at least one existing flow. Additionally, identical data, i.e., data from the same source, may be deduplicated, resulting in even further reduction. The exact impact of those effects in a concrete case is difficult to predict and requires detailed statistics, which are costly at runtime in both computation and communication. We therefore limit ourselves to estimates that can be achieved with reasonable effort as heuristics.

Factors of Utilization Change

For a redirect operation, the following set of conditions impact the change in utilization of the involved nodes:

1. Does S need a new connection to U? A connection between two nodes means that there is some regular communication between the two. This is equivalent to a transport protocol (Layer 4) connection, but also applies if connectionless protocols such as UDP are used. The cost we assume for a connection arises from (i) the regular neighbor state information exchange (cf. Section 6.3) and (ii) the fact that message exchanges introduce packet header overhead. In the ideal case, where all messages on a connection can be aggregated into one, the packet header overhead is almost constant and can therefore be treated as a connection cost.

   If S so far does not send events via U, either directly or to be forwarded, a new connection is needed. In most practical cases, S will only choose U as a forwarder if it already has a connection; otherwise S will most likely not have the information about U necessary for the operation (cf. Section 6.3). If there is no existing connection, a link usage penalty corresponding to the expected additional connection has to be added.

2. Can S remove its connection to D? If after the redirect operation, S has no other event flow that is forwarded via D, this connection can be removed. Consequently, the savings for that connection can be counted as the inverse of the penalty above.

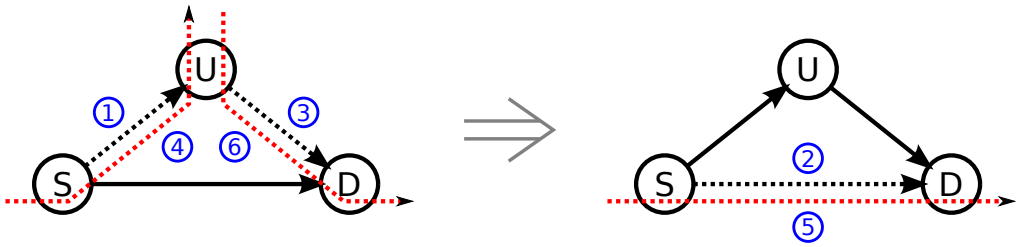


Figure 6.3.: Illustration of the six conditions affecting the expected change in uplink utilization of nodes S and U in the redirect case. The blue numbers correspond to the list items in the text. Black lines are direct connections, red lines are paths that might start and/or end beyond the considered nodes. Dotted lines indicate potential connections/paths, each relevant for one of the items.

3. Does U need a new connection to D? If U did not have any event flow sent via D, it has to create one. This is counted as a penalty like for the first condition. This case is more likely than the first condition, as only the initiator S needs to know U and D. However, for the path latency estimation (cf. Section 6.2.2) the latency between U and D has to be known, requiring a caching of latencies for the occurrence of this case.

4. Does S have an existing flow from the same source via U? The previous conditions considered cases where a connection is used for multiple flows, independently of their properties. If, however, two or more flows from the same source, i.e., with the same content, are routed via the same node, the aggregation potential is much higher than in the above cases. The application payload, i.e., the identical part of the messages, can be completely deduplicated to an almost constant size, independent of the number of receivers. Refer to Section 7.5 for details on the aggregation and deduplication algorithm. Only the individual parts of the messages, such as timing information for latency measurements, remain of linear size with the number of receivers.

   Hence, if S already has a flow from the same source via the new forwarder U, the expected outgoing traffic of S is reduced according to the flow payload in the estimation.

5. Does S have another flow from the same source via D? If S had more than one flow from the same source via the previously next hop node D, the possible savings from the previous item are canceled out. We therefore add the corresponding amount to the traffic expectation.

6. Does U have an existing flow from the same source via D? The same consideration as for node S in Condition 4 can be made for node U. U's outgoing traffic increase from the additional flow is lower if U already has at least one flow from the same source via D. However, the estimation must be made by the initiating node S, for which this information is expensive to obtain (cf. Section 6.3), because it would require detailed information on the current routing configuration of U. We therefore omit this factor for the practical estimation.

The six cases are illustrated in Figure 6.3.

For shortcut operations, analogous considerations are made, but in the inverse effects and from the point of view of the initiating node U:

1. Can S remove its connection to U? Inversely equivalent to Condition 1 for the redirect operation, if S has no other flows via U, it can remove this connection.

2. Does S need a new connection to D? Inversely equivalent to Condition 2 for the redirect operation, S needs a new connection to D, if there was none.

3. Can U remove its connection to D? Inversely equivalent to Condition 3 for the redirect operation, U can remove its connection to D if there are no other flows.

4. Does S have another flow from the same source via U? If U, that after the shortcut will no longer be forwarder for the given flow, forwards other flows from the same source, the savings on the connection from S to U are low due to aggregation and deduplication. This has to be considered as a penalty, inversely to Condition 4 for the redirect operation.

5. Does S have an existing flow from the same source via D? If, in turn, S has other flows from the same source already routed via D, the same amount of traffic is saved. This case, however, cannot be determined cheaply by the initiator U, like in Condition 6 for the redirect case. We therefore omit this factor for the practical estimation. Although this reduces prediction accuracy, we are pessimistic in all cases. Unlike too optimistic approximations, this has the advantage of provoking no oscillations in the optimization process.

6. Does U have another flow from the same source via D? This is the inverse of Condition 6 for the redirect case. In contrast to that, here U can easily determine whether this condition holds. If it does, the expected traffic savings of U are reduced due to the aggregation and deduplication in the current configuration.

6.2.2 Path Latency

For the delivery latency part ($\sum_{(v,u)\in E} U_{\text{lat}}(\mathscr{L}_{u\to v}, I(v,u))$), we need a little further decomposition. As introduced in Section 5.1.1, the overall path latency is modeled as the sum of node-to-node network latencies over all hops and the queuing delays on the nodes. Knowing the full path latency (i.e., end-to-end latency), the effect with respect to network latency can be calculated by adding and subtracting the latencies of the hops that are removed and added, respectively. For a redirect, the initiator S thus has to know about the network latencies to D (the current next hop), to U (the new next hop), and from U to D. Similarly, the initiator of a shortcut, U, has to know the latency from S to itself, from itself to D, and from S to D.

The queuing delay is affected on those nodes along the path where the bandwidth utilization changes, as addressed in the previous section. For queuing delays to be considered on the flow that is modified by the operation, the queuing model from Section 5.4 is used. Hence, the resulting latency prediction is defined as

$$\begin{aligned}\mathscr{L}'_{S\to D} := \mathscr{L}_{S\to D} &- \ell_{\text{net}}(S,D) + \ell_{\text{net}}(S,U) + \ell_{\text{net}}(U,D)\\ &- \ell_{\text{q}}(S) \quad + \ell'_{\text{q}}(S) \quad + \ell'_{\text{q}}(U)\end{aligned} \tag{6.1}$$

for redirect and

$$\begin{aligned}\mathscr{L}'_{S\to D} := \mathscr{L}_{S\to D} \; &+\ell_{\text{net}}(S,D) \; -\ell_{\text{net}}(S,U) \; -\ell_{\text{net}}(U,D) \\ &+\ell'_{\text{q}}(S) \qquad -\ell_{\text{q}}(S) \qquad -\ell_{\text{q}}(U)\end{aligned} \tag{6.2}$$

for shortcut operations, where $\ell_{\text{q}}(v)$ is the current queuing delay estimation for node v and $\ell'_{\text{q}}(v)$ is the estimated queuing delay after the utilization change of v expected from the operation.

6.2.3 Utility Estimation Algorithm

The estimation concepts for link utilization and latencies from the previous two sections can now be put together into a single algorithm that predicts the change in utility (cf. Δu in Algorithm 1, Section 4.4) for a given redirect or shortcut operation. This algorithm is sketched as Algorithm 2.

With regard to the generic local optimization algorithm (Algorithm 1), the function UtilityDelta combines the functions PredictState, PredictMetrics, and Utility to compute Δu directly. The function projects the three main factors affecting the utility change of an operation: the utilization of node S (b'_S), the utilization of node U (b'_U), and the path latency of the the affected flow ($\mathscr{L}'_{flow}$).

Lines 3 and 4 initialize the utilizations for the two nodes S and U with the current values. Then, depending on the operation, the conditions from Section 6.2.1 are checked and the respective utilization modifications are applied. The function Throughput returns the current throughput measurement for the given flow. Utilizations are in all cases modified by adding the additional or subtracting the removed traffic divided by the respective node's link capacity $\mathscr{B}_v$. HasConn(v, u) determines whether v has an active direct connection to u. NumFlows(v, u) returns the number of flows being routed by v via u. NumFlowsFromSourceVia(v, s, u) returns the number of flows from source s that are routed by v via u. UsagePerConnection provides an estimation for the constant traffic factor of a direct connection between two nodes, taking into account the regular node state updates (cf. Section 6.3) and packet header overhead.

Lines 17 and 31 calculate the path latency change according to Section 6.2.2 for redirect and shortcut, respectively. The following lines (18 and 32) project the queuing delays on the involved nodes. The function QueuingDelay estimates the queuing delay (ℓ_{q}) based on the respective node's link utilization, according to the model described in Section 5.4.

Finally, lines 33 and 34, calculate the expected utility difference for link utilization (Δu_{bw}) and path latency (Δu_{lat}), respectively. This is done by taking the difference of the utilities of the predicted values after the operation and the utilities of the current state. Those utilities are then summed up, together with an optional RerouteOffset. This offset is a negative number or zero, allowing to consider rerouting costs as a constant factor, corresponding to u^- in Algorithm 1 (Section 4.4).

Hence, the return value of the function UtilityDelta is the expected difference in the overall utility of the system after applying the given operation op. A negative value indicates that the given operation will degrade the overall system utility and is therefore not worth applying, while a positive value indicates that an improvement is expected from the operation.

Algorithm 2 Algorithm sketch for the prediction of utility change for a given redirect or shortcut operation.

function UTILITYDELTA(op, $flow$, S, U, D)
  $src \leftarrow$ SOURCENODE($flow$) ▷ Flow's source node
  $b'_S \leftarrow b_S$ ▷ Current uplink utilization of S
  $b'_U \leftarrow b_U$ ▷ Current uplink utilization of U

  if $op =$ REDIRECT **then** ▷ **Redirect** operation
    $b'_U \leftarrow b'_U +$ THROUGHPUT($flow$)$/\mathcal{B}_U$ ▷ U will have to forward the flow
    if HASCONN(S, U) **then** ▷ (cf. Section 6.2.1, Redirect Item 1)
      $b'_S \leftarrow b'_S +$ USAGEPERCONNECTION($S, flow$)$/\mathcal{B}_S$
    if NUMFLOWS(S, D) $= 1$ **then** ▷ (cf. Section 6.2.1, Redirect Item 2)
      $b'_S \leftarrow b'_S -$ USAGEPERCONNECTION($S, flow$)$/\mathcal{B}_S$
    if HASCONN(U, D) **then** ▷ (cf. Section 6.2.1, Redirect Item 3)
      $b'_U \leftarrow b'_U +$ USAGEPERCONNECTION($U, flow$)$/\mathcal{B}_U$
    if NUMFLOWSFROMSOURCEVIA(S, src, U) ≥ 1 **then** ▷ (cf. Section 6.2.1, Redirect Item 4)
      $b'_S \leftarrow b'_S -$ THROUGHPUT($flow$)$/\mathcal{B}_S$
    if NUMFLOWSFROMSOURCEVIA(S, src, D) > 1 **then** ▷ (cf. Section 6.2.1, Redirect Item 5)
      $b'_S \leftarrow b'_S +$ THROUGHPUT($flow$)$/\mathcal{B}_S$
    $\mathcal{L}'_{flow} \leftarrow \mathcal{L}_{flow} - \ell_{\text{net}}(S, D) + \ell_{\text{net}}(S, U) + \ell_{\text{net}}(U, D)$ ▷ Routing latency difference
    $\mathcal{L}'_{flow} \leftarrow \mathcal{L}'_{flow} -$ QUEUINGDELAY(b_S) + QUEUINGDELAY(b'_S) + QUEUINGDELAY(b'_U) ▷ Projected queuing delays

  else ▷ **Shortcut** operation
    $b'_U \leftarrow b'_U -$ THROUGHPUT($flow$)$/\mathcal{B}_U$ ▷ U will no longer forward the flow
    if NUMFLOWS(S, U) $= 1$ **then** ▷ (cf. Section 6.2.1, Shortcut Item 1)
      $b'_S \leftarrow b'_S -$ USAGEPERCONNECTION($S, flow$)$/\mathcal{B}_S$
    if HASCONN(S, D) **then** ▷ (cf. Section 6.2.1, Shortcut Item 2)
      $b'_S \leftarrow b'_S +$ USAGEPERCONNECTION($S, flow$)$/\mathcal{B}_S$
    if NUMFLOWS(U, D) $= 1$ **then** ▷ (cf. Section 6.2.1, Shortcut Item 3)
      $b'_U \leftarrow b'_U -$ USAGEPERCONNECTION($U, flow$)$/\mathcal{B}_U$
    if NUMFLOWSFROMSOURCEVIA(S, src, U) > 1 **then** ▷ (cf. Section 6.2.1, Shortcut Item 4)
      $b'_S \leftarrow b'_S +$ THROUGHPUT($flow$)$/\mathcal{B}_S$
    if NUMFLOWSFROMSOURCEVIA(U, src, D) > 1 **then** ▷ (cf. Section 6.2.1, Shortcut Item 6)
      $b'_U \leftarrow b'_U +$ THROUGHPUT($flow$)$/\mathcal{B}_U$
    $\mathcal{L}'_{flow} \leftarrow \mathcal{L}_{flow} + \ell_{\text{net}}(S, D) - \ell_{\text{net}}(S, U) - \ell_{\text{net}}(U, D)$ ▷ Routing latency difference
    $\mathcal{L}'_{flow} \leftarrow \mathcal{L}'_{flow} +$ QUEUINGDELAY(b'_S) $-$ QUEUINGDELAY(b_S) $-$ QUEUINGDELAY(b_U) ▷ Projected queuing delays

  $\Delta u_{bw} \leftarrow U_{\text{bw}}(b'_S) + U_{\text{bw}}(b'_U) - U_{\text{bw}}(b_S) - U_{\text{bw}}(b_U)$
  $\Delta u_{lat} \leftarrow U_{\text{lat}}(\mathcal{L}'_{flow}) - U_{\text{bw}}(\mathcal{L}_{flow})$
  return REROUTEOFFSET() $+ \Delta u_{bw} + \Delta u_{lat}$

6.3 Neighbor Information Exchange

For the above estimations to be computed locally on the initiating node, the necessary information must be available on that node. In the following, we briefly analyze what this information is and what effort it takes to keep it up to date at the respective nodes.

- All connected neighbors, that is, the nodes known as senders and receivers as well as those already used as forwarders, serve as potential forwarders. Of those, *link utilizations and capacities* need to be exchanged. This information is in the order of $O(n)$, where n is the number of neighbors. It can be regularly piggy-backed on data that is sent between the nodes anyway. Since it only needs to be updated when a value changes, its overhead is negligible.

- When selecting forwarders, it is crucial that the forwarder is connected to the next-hop node. Establishing a new connection on demand is possible, but the necessary handshake round-trip prohibits an immediate switch of routes. Therefore, nodes need to be aware of their neighbors' neighbors, i.e., their *two-hop neighborhood*. Exchanging this information naively causes $O(n^2)$ traffic (with n again being the number of neighbors). In addition, this information may change regularly and thus requires regular updates.

  In fact, however, we do not need the whole two-hop neighborhood, but only the set of *common neighbors* in that neighborhood. This is due to the fact that in the redirect case, the old next-hop node, D, is always known to the initiator S. S thus only needs to know whether U is a common neighbor of S and D. Common neighborhood can be exchanged efficiently using bloom filters. U sends the filter with its neighbors to S, which can test its neighbors against the filter. This way, it finds out whether U is connected to D. Bloom filters reduce the complexity of the exchanged information to $O(n)$. False positives in the bloom filter test can lead to S erroneously believing that a potential forwarder knows the next-hop node. In such unlikely case, the selected neighbor just rejects the forwarding request.

- Besides the information whether a potential forwarder knows the next-hop node, it is also relevant whether it has already *existing flows* to that node, as discussed above. Here, the same technique is applied as for the common neighbors, using an additional bloom filter. Hence, the resulting complexity is again $O(n)$.

- For the prediction of path latency changes, the added and removed *network latencies*, $\ell_{\text{net}}(S,D)$, $\ell_{\text{net}}(S,U)$, and $\ell_{\text{net}}(U,D)$, have to be known. The connections adjacent to the initiating node ($\ell_{\text{net}}(S,D)$ and $\ell_{\text{net}}(S,U)$ for redirect and $\ell_{\text{net}}(S,U)$, and $\ell_{\text{net}}(U,D)$ for shortcut) are known to that node by monitoring the respective connections. As discussed in Section 4.2, network latency is assumed to be half the round-trip time, which in turn is measured by piggy-backing timing information onto the packets that are transmitted over the respective connection, thus causing little overhead.

  What remains is the third network latency component, $\ell_{\text{net}}(U,D)$ and $\ell_{\text{net}}(S,D)$, respectively. For having the necessary information for all possible options available, each node

Information	Communication complexity	m in InterestCast	Resulting complexity
Neighbor (ID/existence/count)	$O(n^{m-1})$	1	$O(1)$
Neighbor properties (constant size)	$O(n^m)$	1	$O(n)$
Common neighbors (naïve .. bloom filter)	$O(n^m)$ .. $O(n^{m-1})$	2	$O(n)$
Inter-neighbor properties (latency/weight)	$O(n^m)$	2	$O(n^2)$

Table 6.1.: Overview of approximations for communication complexity of different types of information to be available in m-hop neighborhoods. n is the average number of neighbors per node.

needs the network latencies of all pairs of common neighbors, which is $O(n^2)$ in the worst case, i.e., with an interest graph clustering coefficient of 1. With a lower clustering coefficient, the number of common neighbors and thereby the number of latency values is lowered correspondingly. Note that n is still the number of neighbors, not the total number of nodes. Moreover, in many cases, the network latency rarely varies significantly in short time intervals, so that the update of latencies does not need to happen frequently.

- Finally, the *path end-to-end latencies* of the paths subject to change needs to be known to predict the utility impacts with non-linear utility functions. This information is monitored by the destination node of each path and propagated backwards along the path. This approach involves that the routing tables contain backwards routing information, because all nodes on the path need the information and it is not certain that there is an inverse path with the same intermediate nodes.

Table 6.1 gives a more generic overview of the different types of information that are collected in the nodes' neighborhoods, as discussed above. The second column gives the communication complexity parametrized with the desired neighborhood size in terms of number of hops (m). n is the average number of neighbors per node. The third and fourth columns show the values of m needed for InterestCast and the resulting complexity, respectively.

6.4 Measuring Network Capabilities

Before exchanging the information with their neighbors, the nodes need to obtain it. Each node therefore needs to measure the relevant data: the current link usage and its capacity, the throughput of each individual path that is of relevance for the node, and the latencies to its neighbors.

6.4.1 Link Usage and Capacity

We consider the node's local link as the bottleneck link, i.e., the last mile bandwidth as the limiting factor, as introduced with the model (Section 5.1). The relevant parameters can be represented in different ways, as total link capacity (in bytes per second), link usage (in bytes per second),

relative usage (utilization), and spare capacity (in bytes per second). Having values for two of the four, estimates for the remaining two can be easily calculated. Depending on the particular use case, different variations can be measured.

Measuring or estimating bandwidth is a common networking problem. Hence, there is already a history of research work dealing with it [88, 30, 80], especially for the purpose of flow and congestion control algorithms [88], but also with respect to dedicated tools [133]. Generally, one can distinguish between active probing and passive measurements. Active probing injects traffic into the network, while passive methods use the existing traffic to infer the available bandwidth. Furthermore, different network models can be assumed [147]. The easiest and most direct way of measuring spare capacity is to actively saturate the link and measure the throughput [147, 149]. Especially for latency-sensitive applications, however, this is harmful because saturating the link introduces additional queuing delays. Therefore, less invasive techniques are preferred, such as timing relationships between pairs or trains of packets [95, 133]. The corresponding tools like *nettimer* [95] use packet traces to analyze the inter-arrival times to deduce the link bandwidth.

Alternatively, assuming that the available bandwidth is approximately constant, we can take it as given, either measured externally or specified by the user. With this option, we just need to measure the used bandwidth to get an estimate on the relative usage. Further, if the maximum bandwidth to be used is given, the output rate should be limited. The rate limiter can then be used to directly gain the relative usage, which is the ratio of the time the channel is free. We use this option in our implementation, assuming that the available bandwidth is known a priori. Including one of the existing bandwidth estimators is just an engineering task.

The usage measurements should be reasonably accurate so that the current state and change predictions have the desired precision. With respect to total and spare capacity, the measurements should not tend to overestimation to avoid overloading nodes. Conversely, the link usage should rather be overestimated than underestimated to avoid actual overload situations. Furthermore, it is necessary to find a good trade-off between agility and smoothness of the measurements. Changes should be reflected quickly in the measurement values so that the system can react timely. On the other hand, transient fluctuations should not disturb the system behavior. Hence, we use an exponential averaging low pass filter.

6.4.2 Route Throughput

In addition to the nodes' total throughput, the optimization requires information about the throughput of each individual path (cf. Section 5.1.1) that is of relevance for the respective node. Relevant paths are those that either start at or are routed via the particular node. This information is needed to predict the usage change of the involved nodes for a given redirect or shortcut operation (Section 6.1).

For each relevant path, a node manages a routing table entry, which can also store the throughput-related information. Since it is of interest to provide estimates with respect to both net payload throughput, which is relevant when packets can be aggregated, and gross throughput including headers, which is relevant when packets are sent individually. We therefore measure the packet rate and their sizes separately. Details can be found in Section 7.3.2.

6.4.3 Latency

Network latencies between pairs of nodes are of interest for predicting the effects of path operations on the path latency. Without perfectly synchronized clocks, it is easy to measure round-trip times (RTT), while one-way latencies cannot be exactly determined. Despite some approaches to estimate one-way latencies [73, 37], we assume the latencies to be symmetric and use $RTT/2$ as the one-way latency. Round-trip times are measured by piggy-backing timestamps with the messages, similar to TCP's extension for round-trip measurements [25]. This mechanism requires two 32-bit timestamps, resulting in a total overhead of 8 bytes per packet.

6.5 Utility Functions

The application-defined utility functions are a crucial part of InterestCast. Although in principle freely chosen by the application, their properties influence how well the system can adapt to the application's needs. In this section, we discuss important aspects to consider regarding InterestCast's utility functions and give some examples.

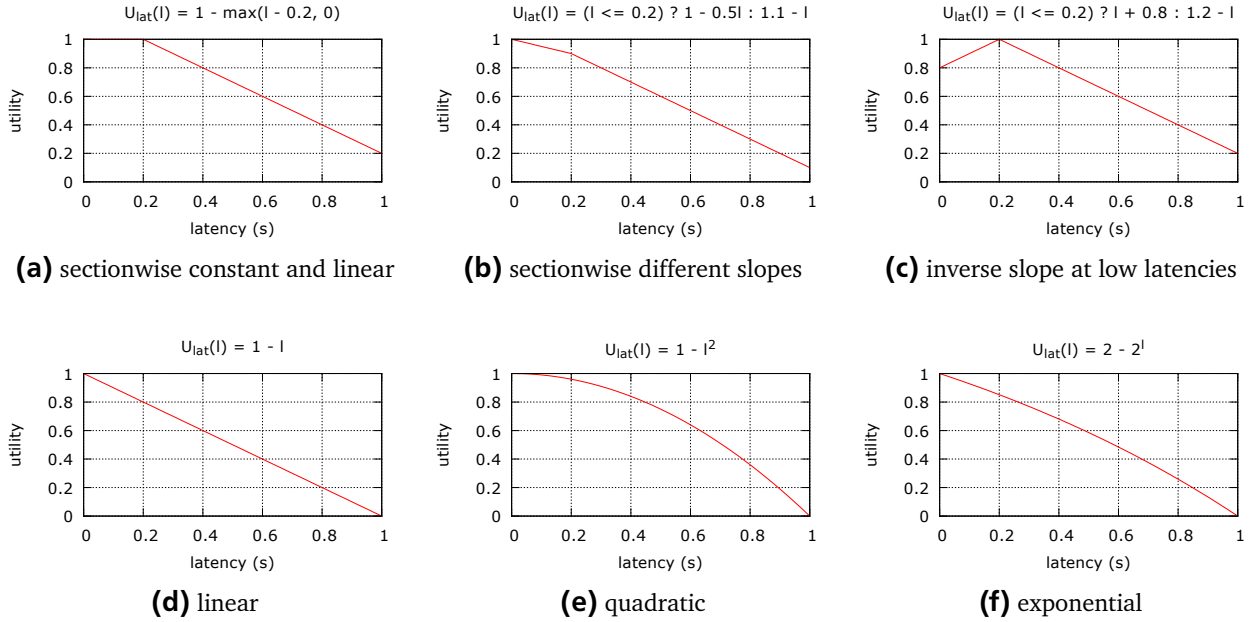
As detailed in Chapter 5, InterestCast uses two utility functions, $\mathcal{U}_{\text{bw}}(b_v)$ for the node utilization utility, and $\mathcal{U}_{\text{lat}}(\mathcal{L}, \iota)$ or $\mathcal{U}_{\text{sta}}(\mathcal{S}, \iota)$ for interest-dependent latency or staleness utility, respectively. Instead of utilities, applications can as well use costs, depending on their performance model. Since we use additive utilities, costs are given as negative utilities.

The two utility functions provided by the application, although in principle freely defined, should fulfill certain properties to be well suited for InterestCast's optimization algorithm:

- They must be monotonic. Further, in areas where the solution is not considered 'good enough', they should be strictly monotonic. InterestCast's greedy local optimization uses their gradient to decide which operation to perform in each iteration individually. If no single operation (redirect or shortcut) changes the utility factors enough to make for a change in utility, no operation is ever performed. It is, however, not necessary for the functions to be convex.

- The utility functions should be defined on an appropriate domain, i.e., in a large enough range. Even though a node's link utilization can in principle not exceed 1, defining th corresponding utility function in a range beyond 1 can be helpful for at least two reasons. First, a node's real link capacity might actually be higher than reported to InterestCast. In such case, the capacity exceedance might be associated with a high penalty, but can sill be achievable. Secondly, even if the capacity cannot be exceeded, the utility estimation can work with a utilization that virtually grows beyond 1 to be able to trade off different 'bad' options against each other. Over-utilized nodes will drop messages, according to the scheduling algorithm (cf. Section 7.4).

- The value range (codomain) of the two utility functions should be comparable in that they allow a trade-off between each other. Based on this, InterestCast decides which amount of utilization reduction is needed to pay off for a certain latency increase or vice versa. The utility weights w_{bw} and w_{lat} can be used to adjust the ratio between the individual functions.



(a) sectionwise constant and linear **(b)** sectionwise different slopes **(c)** inverse slope at low latencies

(d) linear **(e)** quadratic **(f)** exponential

Figure 6.4.: Exemplary latency utility functions

- InterestCast does not consider fairness explicitly, e.g., by evaluating a dedicated fairness function (Jain's fairness index [82]) or by explicitly maximizing the minimum utility among nodes (max-min fairness [20, ch. 6.5.2, pp. 524–529]). By defining the utility functions appropriately, however, the system can be nudged to a fair distribution of performance and load. Generally, this can be achieved by using superlinear cost functions (i.e., superlinear negative utility functions), such as high-degree polynomials. With those, reducing performance for a node that already achieves low performance or increasing load on an already loaded node will be overproportionally costly and the system will tend to reduce other nodes' performance or increase their load, respectively.

To conclude, it is to be noted that the utility functions do not indicate feasibility of a solution, but only its value, therefore assuming every solution is feasible no matter how bad it is. This is due to InterestCast's best-effort guarantees. InterestCast does not have the concept of being 'fully booked'. Furthermore, there is no 'zero' utility, i.e., the utility value 0 has no special meaning. Utilities can be all positive, all negative, or mixed, as long as they are comparable in that maximizing the sum of all utilities maximizes the overall utility. In the following, we use as a convention a utility value of one for the 'ideal' case and zero for 'bad' cases. Yet, utilities can fall to any value below zero, representing the 'worse' cases.

6.5.1 Latency Utility

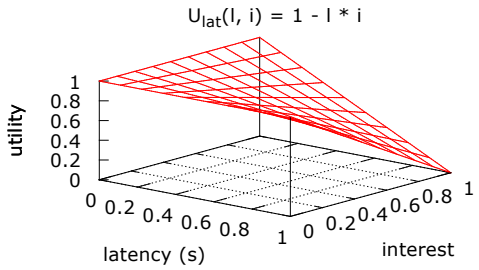
The purpose of the latency (or staleness) utility function is to directly express the application's performance requirements. As discussed in Chapter 2, different applications have different needs with respect to latencies. The effects of latency to user experience have been particularly well studied for the application class of online games [44, 43, 61, 18], see also Section 2.1. We first

consider the exemplary case of 200 ms being the critical limit, beyond which there is a negative impact on the user experience. The corresponding utility function could be a sectionwise defined function, assigning a constant value of one in the interval from 0 to 200 ms and afterwards falling linearly (Figure 6.4a). If the latency is to be minimized even if it is already below the critical value (200 ms in our example), the utility function may be defined to fall gradually in that interval (Figure 6.4b). Conversely, to optimize for fairness, lower latencies could as well be penalized so that the system tends to bring all latencies close to a target latency (Figure 6.4c).

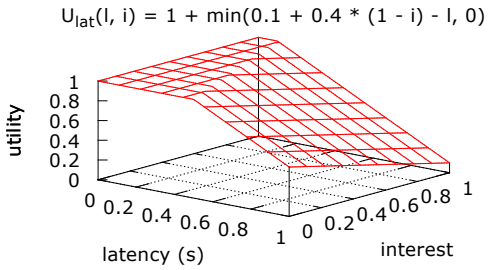
Alternatively, if there is no clear threshold, a very simple utility function could be linear with a negative slope (Figure 6.4d). To achieve better fairness, as discussed above, the function can be defined with superlinear negative growth, e.g., quadratic (Figure 6.4e) or exponential (Figure 6.4f), to increase the penalty for particularly high latencies. Finally, the latency utility function could be sampled directly from measurements of specific applications, such as player performance in online games [43].

InterestCast uses the interest level as a second parameter to the latency utility function, making it bivariate. The interest level parameter allows modifying the utility for a given latency. Typical uses are the variation of the critical limit of latency and the variation of the slope of the utility depending on the latency. For example, the critical latency could range from 100 ms for the highest interest level of one, to one second for an interest level close to zero. Like for the latency, the interest level can be taken into account linearly, as a polynomial, or exponentially. Figure 6.5a shows an example which is linear in both dimensions. A stepwise linear function is shown in Figure 6.5b, where the critical latency threshold is dependent on the interest level. Here, too, a growth superlinear with the latency is desirable if fairness is of concern. Furthermore, it may be desired that the interest level has a sublinear impact; for instance, if the interest level is linear with the distance in a spatial use case, events up to a certain distance may still be of high importance, while the importance falls rapidly close to the vision range borders. An example for a resulting function is shown in Figure 6.5c. Fulfilling all of the desired properties discussed in the beginning of this section, such function appears promising with respect to a good optimization performance. In principle, InterestCast also allows the specification of binary utility functions as exemplified in Figure 6.5d. However, since such functions do not fulfill the desired properties, they do not seem particularly promising.

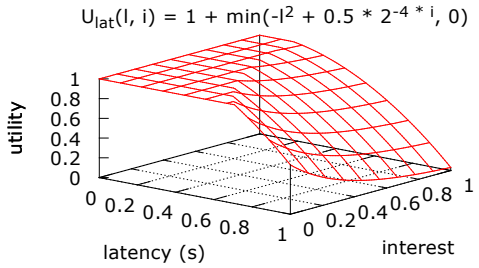
The latency given as input to the utility function is a time value, which has been ignored so far. We assume the unit of time to be seconds and keep omitting this unit in the following. There remains, however, the problem of the normalization of the utility, especially in relation to the link utilization utility to which the latency utility should be comparable. Unlike the latency, the link utilization is already normalized with respect to the node's capacity. This can be solved by normalizing the latency with a reference latency, which can be the critical latency or some other target latency. As before, the target latency can be dependent on the interest level. Figure 6.5e shows an example with a quadratic dependency from the normalized latency. The normalization is linear with the interest level so that a utility of zero is reached at a latency of 100 ms for an interest level of one and 1 s for an interest level of zero.



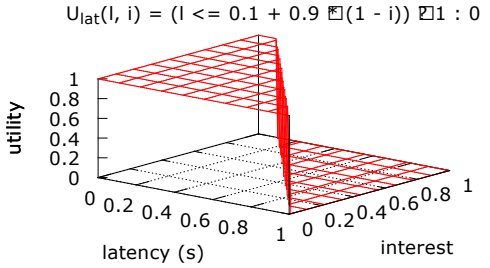
(a) linear in latency and interest, product



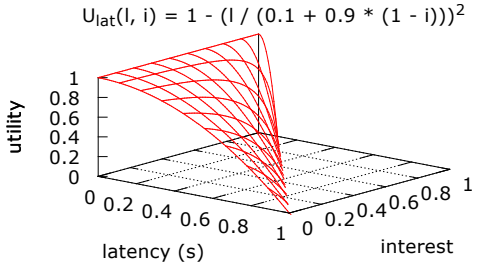
(b) sectionwise linear



(c) sectionwise, quadratic with latency, threshold exponential with interest



(d) binary



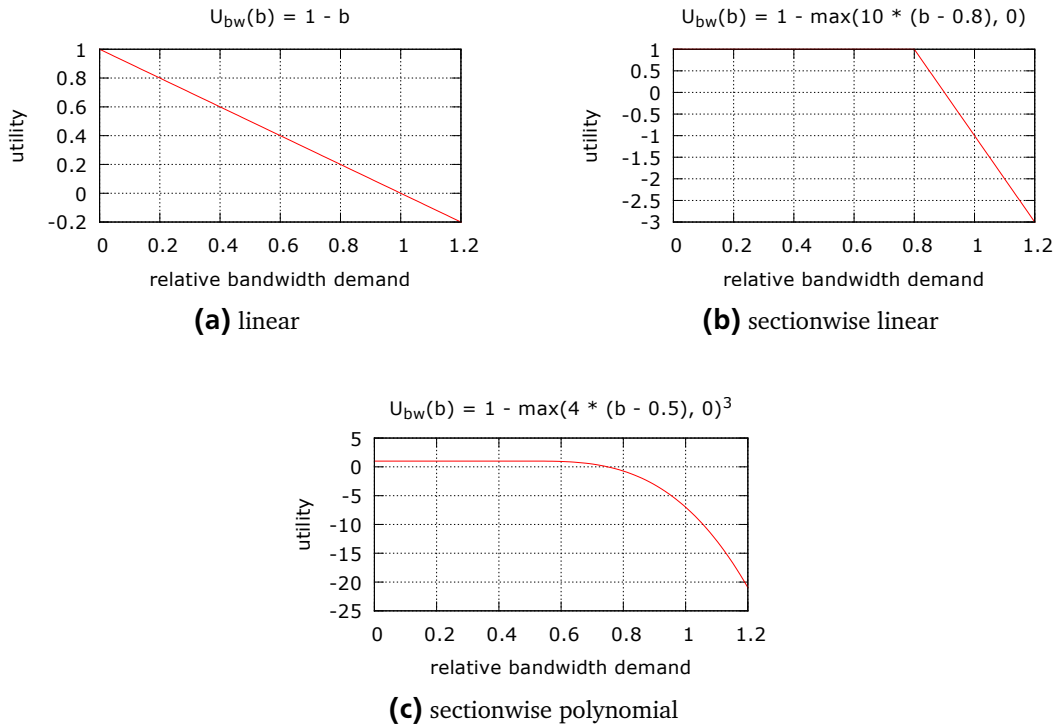
(e) quadratic with latency, normalized linearly with interest

Figure 6.5.: Exemplary bivariate latency-interest utility functions

6.5.2 Bandwidth Demand Utility

The bandwidth demand utility function accounts for the costs induced by the network traffic on the participating nodes. Directly, this is not related to application performance, but it is up to the application developer to decide on the trade-off between performance and cost. Hence, this utility function is to be defined by the application on the same level as the latency utility function.

In addition to the direct use, the bandwidth demand utility function has important indirect purposes. First, although the queuing model (cf. Section 5.4) accounts for utilization-induced delays, it does not consider potential load changes. Expected queuing delays skyrocket when the utilization reaches close to 100%, but remain insignificant below that. However, queuing of instantaneous packets can lead to unwanted spikes in the queuing delay. The queuing model does



(a) linear

(b) sectionwise linear

(c) sectionwise polynomial

Figure 6.6.: Exemplary utility functions for bandwidth demand

not provision spare capacity to cope with those. The bandwidth demand utility function can be used to achieve this by assigning a growing utility penalty to link utilizations going towards a critical point that is located below 100%. Secondly, a superlinear bandwidth demand cost can be used to trim the optimization process towards a more balanced load to improve cost fairness, analogously to the latency fairness.

Figure 6.6 shows some exemplary bandwidth demand utility functions. A simple linear case is given in Figure 6.6a. If we do not assume any costs associated with bandwidth demand but want to penalize high utilizations, we can start reducing utility at a given threshold; Figure 6.6b uses 80%. Figure 6.6c shows a superlinear cost example, starting to decrease at a utilization of 50%.

6.5.3 Utility Weights

As already indicated in Chapter 5, the two utility functions for latency and bandwidth demand are weighted using the factors w_{lat} and w_{bw}, respectively. In detail, there are three main purposes of the weighting factors:

- They are used to fit the utility functions, if those are defined in different codomains. Although the corresponding factors can be included directly in the utility functions, the separation might improve clarity.
- They allow tuning the ratio between the two for influencing the trade-off between bandwidth usage and latency without the need for updating the utility functions as a whole.

- Finally, as already discussed in Section 5.5, they compensate the different amount of nodes versus paths and therefore the varying weights of their utility sums.

In this section we have shown and discussed several possible utility function designs. Utility functions allow specifying optimization goals directly based on application needs on a best-effort basis. However, as we have seen especially for the bandwidth demand utility functions, there remains room and the need for engineering when defining the functions, including the consideration of influencing factors and balancing trade-offs.

7 Prototype

This chapter describes the prototypical implementation of the InterestCast algorithm. We first introduce the high-level architecture describing the integration of InterestCast in the overall application. Afterwards, we provide an overview of the InterestCast internals, following its internal protocol layering. We then detail the important parts of the InterestCast implementation, which go beyond the core algorithm that is discussed in Chapter 6. The implementation is integrated in the Planet PI4 evaluation platform, which is described in Chapter 8.

7.1 High-Level Architecture

As discussed in Section 4.1.1, InterestCast is designed to work underneath an interest management component, such as pSense. The interest management component is responsible for detecting new neighbors, i.e., participants of interest, and passing them to InterestCast. InterestCast does not bring its own neighbor discovery function, because this cannot be implemented in a fully application-agnostic way. Therefore, this is part of the interest management. InterestCast, however, manages the neighbors once they are connected and terminates obsolete connections. The interest management is furthermore responsible for setting InterestCast's interest levels. Optionally, it may also give predictions or estimates for the interest of neighbors. This allows to react faster to interest changes and the addition of new neighbors. The interest management module passes application-level messages to InterestCast for their dissemination. It can use InterestCast's messaging functionality for its own management purposes, but does not have to. Figure 7.1 illustrates this basic interaction.

7.2 InterestCast Components and Layers

Internally, InterestCast is split into several sub-components, which are illustrated in Figure 7.2. The components are aligned as a layered stack. The top components provide the interfaces for the upper layers, including the respective functions for message dissemination and interest management. The end-to-end layer provides messaging along the routing paths and end-to-end information such as latencies. Routing and optimization interact closely in that the optimization component initiates route updates. The hop layer keeps information on directly connected neighbors, which is used mainly for the optimization process. Scheduling manages the outgoing message queue and creates batches of messages for the same node to be aggregated and deduplicated by the aggregation component, where possible. Finally, the connection management is a thin layer handling connections to neighboring nodes and abstracting from the transport layer.

InterestCast's internal component layering determines the internal message flow. Messages are propagated downwards and upwards along the stack when being sent and received, respectively. This process is illustrated in Figure 7.3. Generally, the upper layers use the lower layers, but there

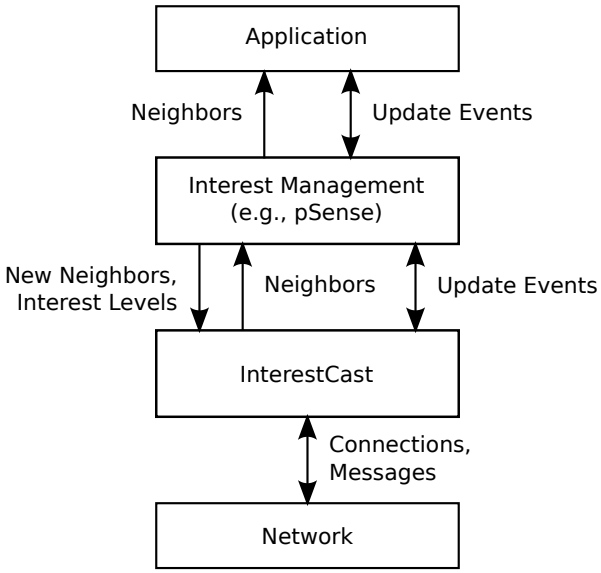


Figure 7.1.: Illustration of the main interaction aspects between the high-level components of interest management and InterestCast as the event dissemination and their surroundings, application and network.

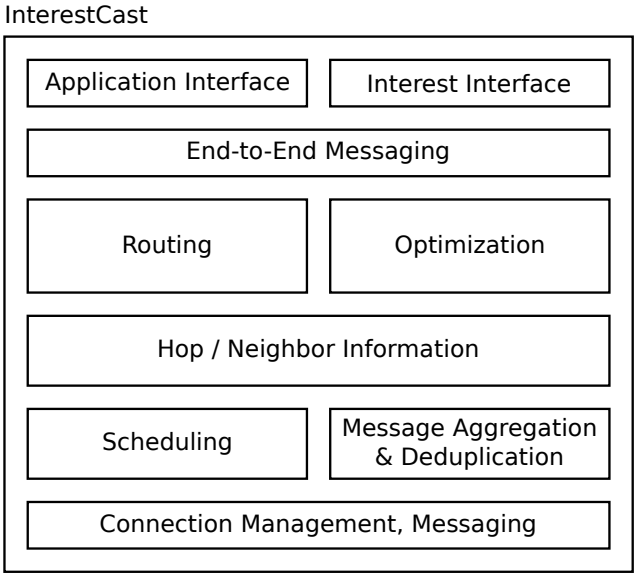


Figure 7.2.: InterestCast's internal components. The thickness of the boxes indicates the complexity of the components.

are exceptions. Layers can be skipped for certain messages and functionality of layers further down the stack can be used directly. While being propagated downwards, messages are kept as a linked list of header blocks, allowing the aggregation and deduplication algorithm to efficiently remove duplicate parts.

In the following, we briefly introduce the individual layers. More detailed discussions on important aspects related to the respective layers are continued in the following sections.

Application Layer

InterestCast's application layer is a thin layer that primarily encapsulates application messages, i.e., all messages that originate from a layer above InterestCast. It also multiplies messages to be disseminated and passes them to the lower layer as individual messages for each recipient. This allows processing, first and foremost the routing, to work on individual messages instead of sets that have to be split according to different routing rules. Before going on the line, redundant messages using the same connection are later deduplicated by the aggregation layer. In addition, the automatic pull of update messages from the application is done by this layer.

Interest Layer

The interest layer is responsible for managing the interest levels of individual neighbors. The interest level of one participant in another can be determined in two ways: explicitly by the interested participant (interest subscription) and implicitly by the participant of interest using interest estimation based on application-specific information such as virtual world positions. This layer integrates both sources, depending on what is provided by the application.

End-to-End Layer

The end-to-end layer encapsulates and monitors messages that are sent via possibly multiple hops from source to destination via the routing layer. The monitored information includes general activity (send, receive) of the respective neighbor, end-to-end latency, and jitter (i.e., the variation of latency over time). This information is used by the optimizer to determine the utility of the used paths. Furthermore, end-to-end path timeouts are detected on this layer, which trigger a fallback to the direct route on the routing layer.

Optimization Layer

The optimization algorithm described in Section 6.1 is run in this component. It uses the information collected by the routing, hop, and scheduling layers and regularly runs an iteration of the optimization algorithm. Route update decisions are passed to the routing layer for execution.

Routing Layer

The routing layer is one of InterestCast's core elements. It manages the routing tables including the monitoring of path information as well as the re-routing operations. The routing determines the path an event message takes from source to destination. Routing decisions are taken locally based on the routing table, which is updated by the re-routing operations. In addition to the forward routing, messages can also be routed backwards along a path. This is needed for path information

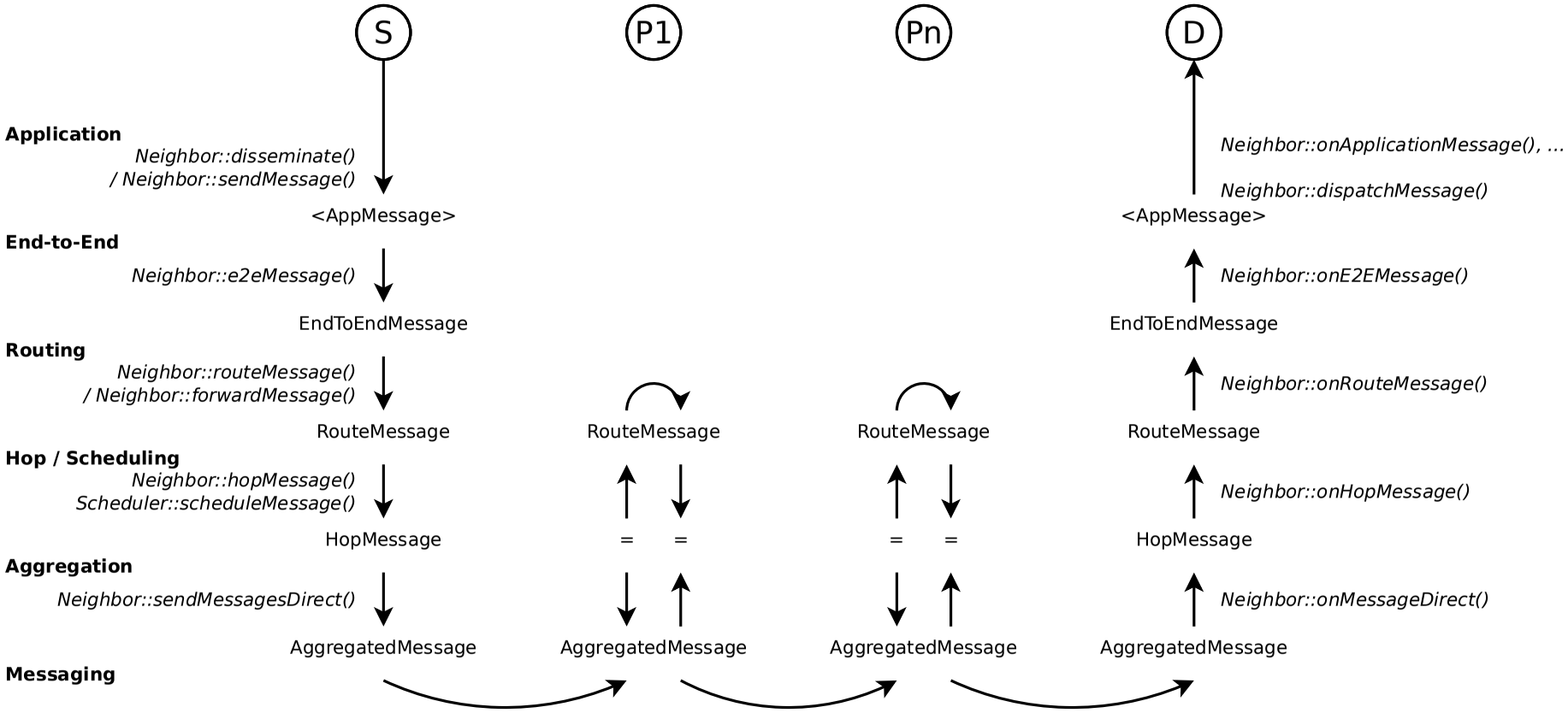


Figure 7.3.: A typical multi-hop message flow through InterestCast's layers from source S to destination D via P1 and Pn.

that is collected on the destination node and needed by all nodes on the path. Since paths are not necessarily symmetric, the regular routing in the opposite direction cannot be used for this purpose. The path information is used by the optimizer. In addition, the data throughput for each route is measured, serving as an additional input to the optimizer. Details on the routing process are explained in Section 7.3, and the route update operations are discussed in Section 7.3.1.

Hop Layer

The hop layer collects and manages information of direct neighbors, i.e., those with an active direct connection. This includes the latencies to the direct neighbors and their activity, i.e., the last interaction in terms of sending and receiving messages. Furthermore, direct neighbors regularly exchange the fundamental information about each other that they need for the local optimization (cf. Section 6.2): the node's link capacity and its usage, directly connected neighbors, the subset of active direct neighbors, and the latencies to the common neighbors.

Scheduling Layer

The scheduling layer is responsible for limiting the outgoing throughput to the amount set according to the node's link capacity or by other use case requirements. Besides the ability to limit bandwidth to a rate lower than actually supported by the node, the primary purpose of the scheduler is to control queuing delays. Instead of having outgoing messages possibly buffered by the operating system's network stack, the scheduler limits the outgoing throughput to a rate that is expected to be sent without significant buffering delay. The second important task of the scheduler is to collect bunches of messages to be forwarded to the same node, so that the aggregation layer can combine them. More details are provided in Section 7.4.

Aggregation Layer

The aggregation layer takes multiple messages to be transmitted to the same node and merges them into a single message where possible. Individual data elements are simply concatenated, while redundant data is deduplicated. The algorithm doing this is presented in Section 7.5. Incoming messages are disassembled into possibly multiple individual messages, which are then passed up the stack.

Connection and Messaging Layer

The connection and messaging layer is a thin layer providing basic connection management, such as initiating and terminating direct connections between pairs of nodes. It provides a messaging interface, abstracting from the actual transport protocol. InterestCast supports both reliable transport protocols like TCP or CUSP [155] and unreliable protocols, most notably UDP.

7.3 Routing

InterestCast's routing is designed with the goal of a low message header overhead. A route in InterestCast is uniquely identified by the (source, destination) pair. Source and destination are thereby identified by their node ID, typically a 64-bit integer. Each end-to-end message needs to be

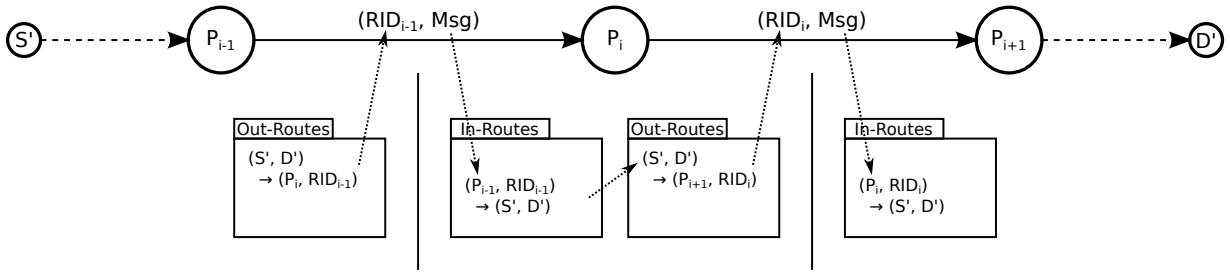


Figure 7.4.: Illustration of the routing process of an event message from source S' to destination D' between the nodes P_{i-1}, P_i, and P_{i+1}. The outgoing routing table (Out-Routes) is used to obtain the route ID (RID) for the successor (via) node. That node uses its incoming routing table (In-Routes) to get the (source, destination) pair identifying the path for further processing.

tagged with routing information so that a forwarder can process it accordingly. In the simplest case, this information is the aforementioned (source, destination) pair. With 64-bit node IDs, this sums up to 128 bits—a non-negligible overhead for small messages. InterestCast therefore introduces a node-local 16-bit *route ID* that uniquely identifies a route. Node-local means that a given ID is valid on the node that stores the respective routing table entry. Therefore, on each hop, a separate route ID is generated and only shared between the sender and receiver for this hop. Even though this approach complicates routing and requires synchronizing more state among the nodes, it saves for small payloads a significant amount of space in the routing header, which is included in every message.

Routing Tables

Each node therefore has two routing tables: one for incoming routes and one for outgoing routes. The outgoing routing table maps from (source, destination) pairs to the via node (i.e., the route successor), its route ID, and further path metadata. The path metadata is propagated back along the path by the destination node and contains the number of hops, end-to-end delay and jitter, and the interest of the receiver. Furthermore, the current message throughput is recorded for each route and stored in the routing table. Finally, if there is a pending route update operation for the route, it is stored so that conflicting update requests can be detected and rejected (cf. Section 7.3.1).

The incoming routing table maps (predecessor, route ID) pairs to (source, destination) pairs, which in turn identify the routing information for further processing.

Message Forwarding

Figure 7.4 illustrates the message routing process among multiple nodes. For the given route from source S' to destination D' identified by the tuple (S', D'), P_{i-1} looks up the next node on the path, P_i, and the corresponding route ID, RID_{i-1}. Together with the route ID, the message is then forwarded to P_i. P_i looks up the route information (S', D') in its incoming routes table based on the

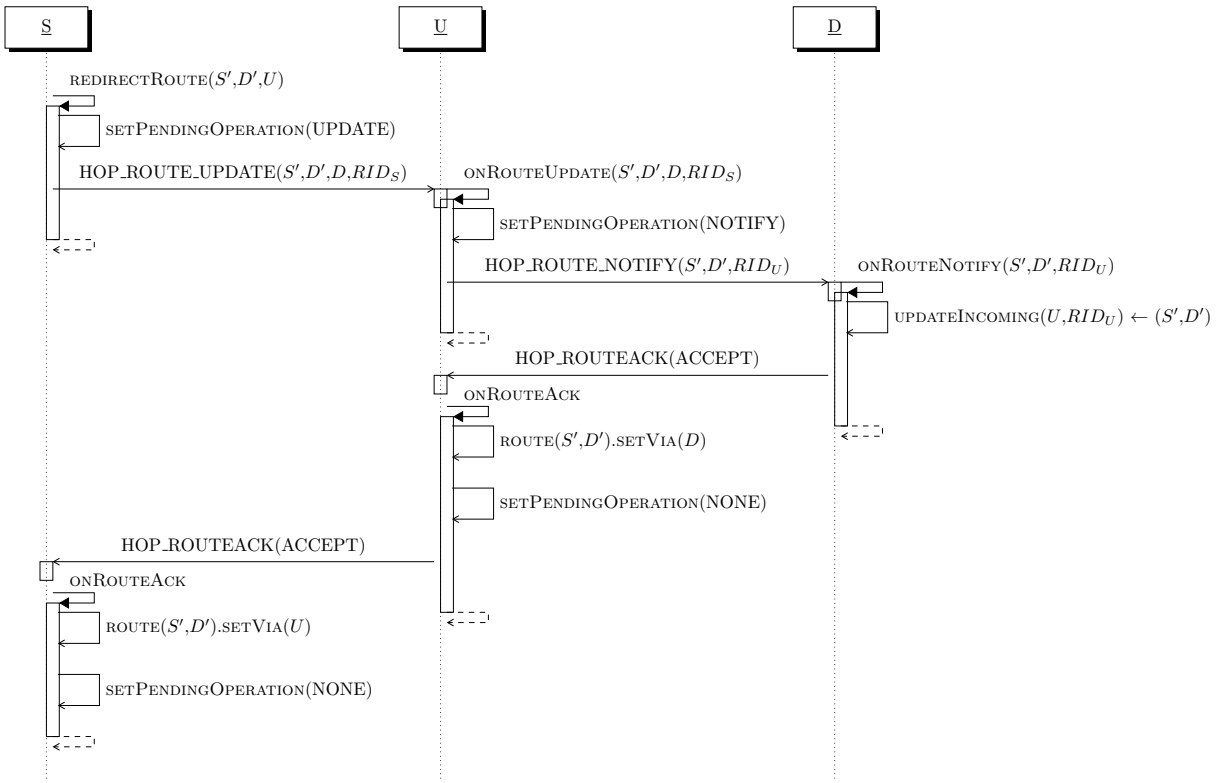


Figure 7.5.: Sequence diagram of a successful route redirect operation initiated by node S, where node U is inserted into the path from source S' to destination D' between the nodes S and D. Note that the path may start before S, i.e., the source S' may be different from S, and the path may end after D, i.e., the destination D' may be different from D. Nodes other than S, U, and D, however, are not directly involved in the operation.

previous node P_{i-1} and the route ID RID_{i-1}. Using the route information, it continues the routing process as described for P_{i-1}.

A special case are direct routes without an intermediate forwarder. In this case, a special value for the route ID is used, indicating that the previous node is the source node.

7.3.1 Route Operations

The reduced packet overhead due to the omission of source-destination pairs in the messages comes with the trade-off of more complex route update processes for redirect and shortcut operations. Those operations are described in this section.

Figure 7.5 illustrates a redirect operation for a route from source S' to destination D' as a sequence diagram. Following the naming conventions from Chapter 6, the initiating node is named S. S may be the source (i.e., S = S') or any node on the path from S' to D', except the destination D'. Initially, S forwards the messages for the given route to D. With the redirect operation, U is inserted between S and D (cf. Figure 6.1).

The basic node interaction works as follows. First, S sends a HOP_ROUTE_UPDATE request message to U, containing the route identification (S', D'), new next-hop node for U, which is D, and the route identifier RID_S that S will use for this route in the following. Consequently, U sends

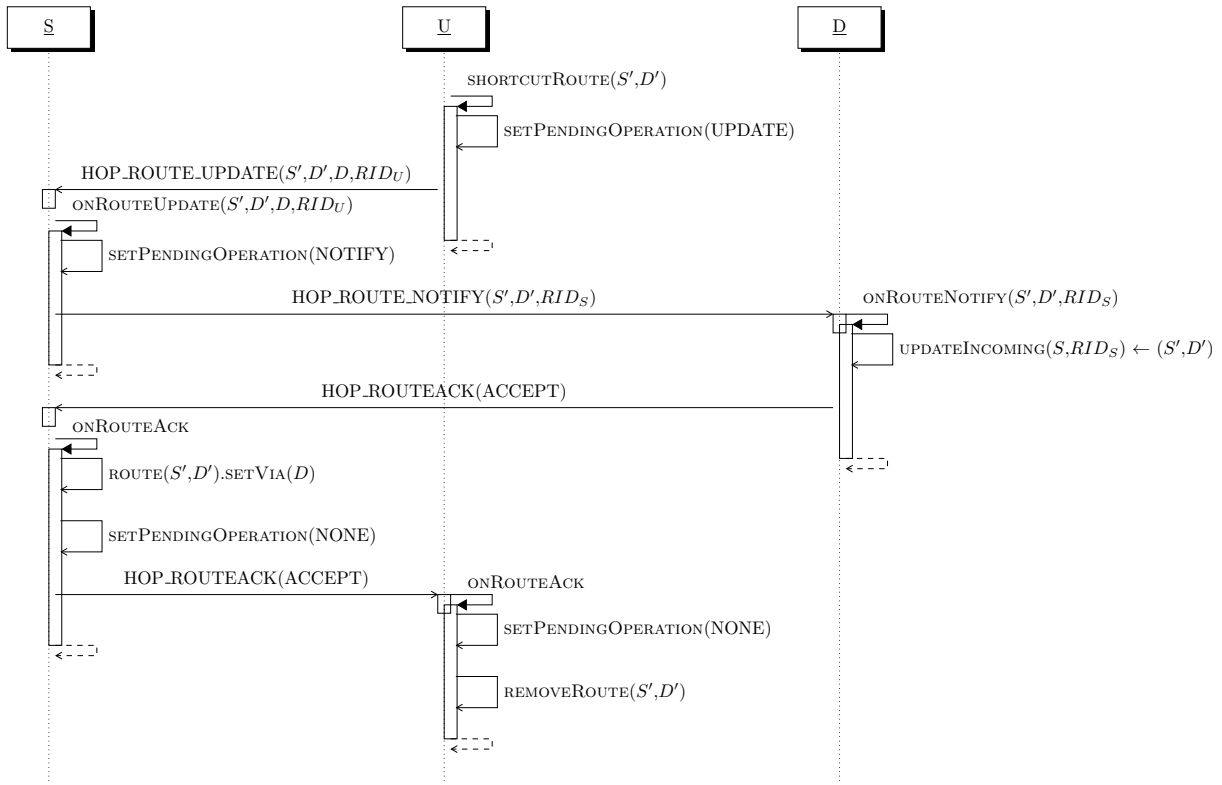


Figure 7.6.: Sequence diagram of a successful route shortcut operation initiated by node U, where node U is removed from the path from source S' to destination D' between the nodes S and D.

a HOP_ROUTE_NOTIFY to D to notify the takeover from S. This message contains the route ID RID_U that U will use for sending messages on the corresponding route to D. D adjusts its incoming route and responds with a HOP_ROUTEACK(ACCEPT). Upon receiving this message, U updates its routing table and in turn responds to S with a HOP_ROUTEACK(ACCEPT).

A shortcut operation works analogously, but is initiated by node U, as shown in Figure 7.6. U sends a HOP_ROUTE_UPDATE to its predecessor on the route, S. S sends a HOP_ROUTE_NOTIFY to the successor D, which updates its incoming route. If successful, S updates its routing information by setting D as the new forwarder for the given route. Finally, U can remove the routing table entry as it is no longer on the corresponding path.

Since the whole operation takes some time, it is necessary to prevent concurrent updates for the same route initiated by other nodes. Such colliding updates would result in inconsistent routing states. Nodes waiting for a response therefore register the pending operation using the SETPENDINGOPERATION function. If a pending operation is set, any further request that affects the respective route will be refused[1]. The entry for the pending operation additionally stores metadata that is used to handle responses for the same operation.

[1] Due to the large number of paths and thus re-routing options and due to the usually short update completion time compared to the update frequency, such collisions are expected to be rare. Yet, they have to be detected to prevent routing inconsistencies.

Route Update Failures

Besides a pending operation for the same route, nodes may reject route operations requested using HOP_ROUTE_UPDATE for several reasons. Those include:

Unknown node If the request asks for forwarding via a node that is not known to the forwarder, it cannot fulfill the request. Generally, the nodes keep each other up-to-date about their neighbors. Therefore, this only happens if the respective neighbor has recently disconnected, and the requesting node has not received the update yet.

Bandwidth exceeded Forwarding requests can be denied if the forwarder decides that it does not have enough spare capacity to handle the additional load.

Already in route If a node that is asked to become forwarder finds out that it is already in the path, it rejects to prevent a routing loop. The necessary information to anticipate this case is not available at the initiating node.

Via reports error This status is reported when the HOP_ROUTE_NOTIFY has been rejected.

HOP_ROUTE_NOTIFY requests have only one fail state:

Unknown route The route to be modified is unknown. This indicates a routing inconsistency and should not happen in normal operation.

Appendix B provides additional flowcharts detailing the operations for handling route update messages.

To prevent an immediate repetition of a rejected route update request in the next iteration of the optimizer, each node stores penalties for routes and neighboring nodes. Depending on the fail state, different penalty values are added to the corresponding route and/or neighbor. The incremental optimizer subtracts the penalties from the estimated utility deltas when evaluating the possible operations (cf. Section 6.1). Therefore operations affecting penalized routes or nodes are weakened. Penalties have an exponential decay so that failed operations can be repeated after some time.

7.3.2 Route Measurements

As identified in Section 6.4, the incremental optimization algorithm needs an up-to-date throughput measurement on each route. Therefore, each node monitors all routes on which it sends data. Upon route operations (cf. Section 6.1) and interest changes, a node regularly gets new routes. Therefore, for route measurements in particular, an important requirement is the prompt availability of measurement data, besides freshness for a quick reactivity and little fluctuation to avoid misinterpretation.

The current throughput of each route is calculated based on the packets that are sent on the respective route, smoothed using an exponential moving average. In its basic form, the exponential moving average calculates a new average m_i for each new sample x_i with $m_i := \alpha \cdot m_{i-1} + (1-\alpha) \cdot x_i$, where $\alpha \in [0,1]$ is the smoothing factor.

Since new routes are created frequently, it is important to obtain a well smoothed value as soon as possible. The exponential moving average requires a special case for the first value m_0. Setting $m_0 := x_0$ exaggerates the value of x_0 for $\alpha > 0.5$. Therefore, a common practice is to start with a simple average and to switch to the exponential moving average after a few samples. The simple average can be calculated in the same way as the exponential moving average, by just replacing the constant α with $\alpha_i' := 1 - \frac{1}{i}$ (which converges to 1 for $i \to \infty$). Instead of switching to the exponential moving average after a certain number of samples, we define $\alpha_i := \alpha_\infty \cdot (1 - \frac{1}{i+1})$, which converges to α_∞ for $i \to \infty$.

While the above calculations assume a constant sampling rate, we can calculate a new throughput sample for each transferred packet. Hence, we adjust the smoothing factor by the inter-arrival time Δt of the corresponding packet (i.e., sample): $\alpha_i(\Delta t) = {\alpha_i}^{\Delta t}$, where Δt is defined as $t_i - t_{i-1}$, with t_i being the arrival time of packet i (in seconds). Putting all together, we get

$$m_i := \alpha_i \cdot m_{i-1} + (1 - \alpha_i) \cdot x_i, \tag{7.1}$$

$$\alpha_i := \left(\alpha_\infty \cdot \left(1 - \frac{1}{i+1}\right)\right)^{t_i - t_{i-1}}. \tag{7.2}$$

To be able to provide estimates on both net and gross throughput, i.e., with and without additional packet headers, the presented measurement method is used for packet size and packet rate separately. The averaged throughput can then be calculated as the product of average rate and size.

7.4 Scheduling

InterestCast's scheduling has two main purposes. First, it limits the outgoing packet rate so that it can perform its own queue management and avoid excessive buffering by the operating system or network elements. Secondly, it collects bunches of messages to be passed to the aggregation at once to be combined and compressed.

The scheduler therefore manages its own queue, a priority queue based on message deadline, message type, and receiver interest level. This way, it prioritizes important and time-critical messages in the case of congestion. The output rate is limited using a token bucket limiter.

The queue further has the ability to dequeue arbitrary messages. Whenever a message is sent out, the scheduler checks the queue for further messages directed to the same neighbor. Those are then piggy-backed to the original message and passed to the aggregation at once. Piggy-backing multiple small messages therefore saves bandwidth due to reduced packet header overhead and due to the removal of duplicate data in the packets.

If the arrival rate of data to be transmitted exceeds the link capacity, i.e., in the case of link congestion, messages need to be dropped to avoid an infinitely growing queue. Conventionally, packets are dropped from the tail of the queue, which is a reasonable option if no further information about the packets is available. In the case of update messages, however, for instance position updates, an update message may be invalidated by a consequent message with more recent information. In such situation, it is undesirable to drop the new message due to a full queue. Instead, the new update should replace the outdated one. This is achieved by assigning update messages to

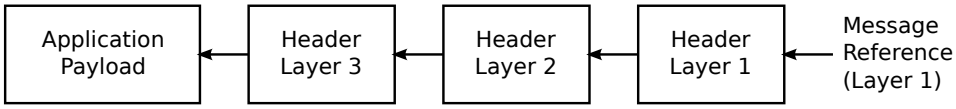


Figure 7.7.: Illustration of a chained message. Each header is added as a separate block, pointing to the following block.

update slots, which indicate that messages in the same slot invalidate previous ones. The deadline-based scheduling preserves fairness. When replacing an update message, the new message inherits the old one's deadline so that it gets a chance to be transmitted earlier.

7.5 Aggregation and Deduplication

As motivated in section Section 4.3, InterestCast treats each flow from one source to one destination separately on the routing and optimization level. Therefore, to achieve bandwidth savings, if multiple messages are going out via the same neighbor node at the same time, they are aggregated. It is the task of the scheduler to collect as many messages that can be aggregated and to push them to the aggregation component at once.

There are two classes of aggregation, which are commonly handled by InterestCast's aggregation algorithm:

1. To save packet headers, multiple distinct messages are bundled into one. This corresponds to a simple payload concatenation.

2. Event messages from the same source typically contain parts that are identical independent from the destination. Such identical parts are stored only once and thus deduplicated.

A simple option to achieve this goal is to concatenate all messages and to use a standard compression algorithm such as Lempel-Ziv [169] or one of its descendants to remove redundant information. Using InterestCast's message structure, however, we propose an algorithm that directly identifies identical blocks to store them only once. For this purpose, messages that are passed down InterestCast's network stack and augmented with additional headers are kept as a chain of message parts. Each additional header is added as a new part. A single message is represented as a linked list of data blocks, see Figure 7.7. If one message is to be disseminated to multiple nodes—which is a common case—, several chains are built using the identical application payload block. This is exemplified in Figure 7.8. The resulting data structure is like a tree, but with reversed edges.

Given a set of message references ('Message A', 'Message B', and 'Message C' in Figure 7.8), InterestCast's aggregation algorithm finds the roots of the inverse message part trees and builds the serialized message based on those trees. Algorithm 3 sketches this algorithm. The two core parts of the algorithm are invoked in Lines 4 and 5. Using the recursive TreeInsert function, all message parts of the messages in M are inserted into the multimap T, which serves as the lookup for children of the non-inverse message part tree. All root message parts (i.e., typically the message payload blocks) are handled as children of the common virtual root node $\emptyset$ (Line 12). The

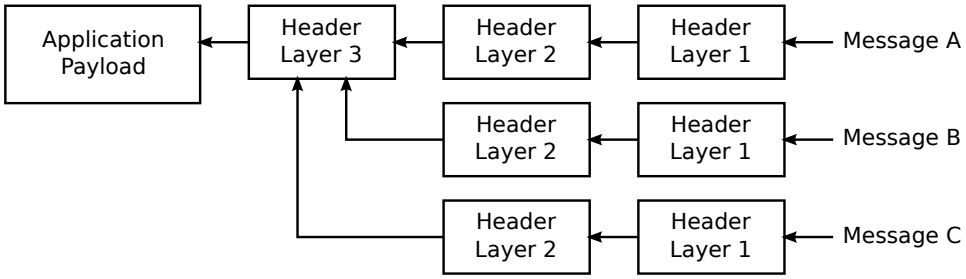


Figure 7.8.: Exemplary messages with identical payload and Layer 3 header, but separate Layer 2 and Layer 1 headers for different receivers.

InterestCast Layer	Header fields	bytes	Header size (bytes)
End-to-End	Message type	1	1
Routing	Route ID	2	6
	Hop count	2	
	Accumulated latency	2	
Hop	Type	1	1
Aggregation	Tag	2*	2*
Messaging	Message size	2	10
	Send timestamp	4	
	Timestamp delta	4	

* per individual message part in aggregate

Table 7.1.: Overview of InterestCast's layer headers and their sizes

SERIALIZE and SERIALIZEBRANCH functions recurse over this tree, starting from the root, to assemble the serialized message in buffer B. Message parts with a common parent in the tree, i.e., siblings, are stored as *sequences*. In the serialized form, message parts are separated with 16-bit *tag* fields, which encode the length of the following block as well as markers for sequence start and sequence end. A sequence start marker indicates that the following parts are a sequence of children, which is terminated by an end marker. Single child cases (e.g., 'Header Layer 1' to 'Header Layer 2' in Figure 7.8) are handled separately. To economize space needed for tags, they are concatenated into a single block (Line 19). The serialization works in two intertwined phases. The SERIALIZE function serializes the actual data of the message parts using the SERIALIZEDATA function, which is omitted here for reasons of clarity. Message sequences are recursed by the SERIALIZEBRANCH function.

7.6 Protocol Headers

Table 7.1 shows an overview of the message headers of InterestCast's layers. The lowest layer, messaging, is not particular to InterestCast and used in a similar manner in the plain pSense configuration. The timestamp fields allow for a round-trip time estimation. The aggregation layer uses two bytes for each message part that is potentially deduplicated. The hop layer only adds

Algorithm 3 InterestCast's message packing algorithm

```
1:  M ← the set of message references to be packed
2:  T ← empty multimap                ▷ T stores the tree of message parts as parent to child mappings
3:  B ← pointer to the destination buffer for the packed message
4:  for m ∈ M do TreeInsert(m)                                  ▷ build tree from all messages in set M
5:  SerializeBranch(∅, B)      ▷ pack message (root is branch, go directly to branch serialization)
6:  procedure TreeInsert(m)
7:      (data, next) ← m      ▷ message m has a data block and optional reference to chained msg.
8:      if next ≠ ∅ then
9:          T[next] ← T[next] ∪ {m}                             ▷ insert as 'child' of next message
10:         TreeInsert(next)                                    ▷ recurse with next message
11:     else
12:         T[∅] ← T[∅] ∪ {m}                                   ▷ insert as root node ('child' of ∅)
13: function Serialize(m, b)                ▷ 1st phase message serialization (header, sequence)
14:     C ← T[m]                                  ▷ get all children for message m from multimap
15:     if C ≠ ∅ then
16:         size ← SeralizeData(m, b)                  ▷ no children → serialize message payload
17:         return(size, False)
18:     if |C| = 1 then
19:         (size1, branchNext) ← Serialize(c ∈ C, b)          ▷ exactly one child → concatenate
20:         size2 ← SeralizeData(m, b + size1)                 ▷ serialize message payload
21:         return(size1 + size2, branchNext)
22:     size ← SeralizeData(m, b)                ▷ multiple children (or root) → serialize payload
23:     return(size, True)                                      ▷ mark as branch-next
24: function SerializeBranch(m, b)          ▷ 2nd phase message serialization (branching)
25:     C ← T[m]                                  ▷ get all children for message m from multimap
26:     if C ≠ ∅ then
27:         return b                                   ▷ no children → nothing to do here
28:     if |C| = 1 then
29:         return SerializeBranch(c ∈ C, b)           ▷ exactly one child, go ahead with this one
30:     for c ∈ C do                                       ▷ more than one child → branch
31:         pTag ← b                  ▷ copy current buffer position for later tag assignment
32:         b ← b + TagSize                ▷ increment buffer pointer to create space for tag
33:         (size, branchNext) ← Serialize(c, b)                        ▷ 1st phase recursion
34:         b ← b + size                          ▷ increment buffer pointer by used size
35:         TagSize(∗pTag, size)                        ▷ assign tag field with block size
36:         if branchNext then          ▷ if next block is branch, mark sequence start in tag
37:             TagSeqStart(∗pTag)
38:         SerializeBranch(c, b)                                   ▷ 2nd phase recursion
39:     TagSeqEnd(∗pTag)                    ▷ end of branch, mark sequence end in tag
40:     return b                                    ▷ return current buffer position
```

a type field to distinguish various hop message types and routed multi-hop messages. The routing layer needs a two-byte route ID whose function is detailed in Section 7.3 as well as a hop counter and latency accumulator for determining path information. Finally, the end-to-end layer adds another type field to distinguish end-to-end message types, such as application messages and InterestCast-internal end-to-end messages.

8 Evaluation Platform

In this chapter, we describe the software platform in which the InterestCast prototype was implemented and tested. This includes, first and foremost, the prototype multiplayer online game *Planet PI4*, the network simulation environment, and experiment data model.

8.1 The Game Planet PI4

Planet PI4 is a third person spaceship shooter game. Each player navigates her ship through a three dimensional space. The game is set in an asteroid field, with the solid asteroids providing some structure in the space. Players join teams and play against the other team(s). The goals are to destroy the other teams' ships and to capture bases. Once a ship is destroyed, the player respawns with a new ship at the team's initial position. Bases are one among different types of strategic points of interest and provide the possessing team with extra energy. To capture a base, it is necessary to stay within the range of the base for an amount of time while keeping players of other teams out. Other points of interest are recovery areas in which players can re-gain health points. Figure 8.1 shows a screenshot of the game.

Planet PI4 was originally developed at the Lehrstuhl für Praktische Informatik IV at the University of Mannheim by Tonio Triebel [159] and has its origins in the SpoVNet project[1]. The game was substantially extended during the QuaP2P project[2] in cooperation of the Technische Universität Darmstadt and the University of Mannheim [102, 100] as well as for the work described in this thesis.

Game World

The asteroid field determines the three-dimensional gameplay region. The region does not have any explicit borders, but there is no incentive to move outside the asteroid field. Bases particularly attract players, causing hotspots in player density. To evaluate scalability, the game map size and player density must be variable. Therefore, the map is generated algorithmically, based on a common random seed. The space is split into regions, so that only the region in which the player is located and the immediately surrounding regions need to be generated and maintained. This further increases scalability in the world size.

8.2 The Planet PI4 Evaluation Platform

The software architecture of the Planet PI4 evaluation platform was designed with the explicit goal of an easy interchangeability of the main components, allowing for testing different compositions of networking implementations under different workloads and network models.

[1] Spontaneous Virtual Networks (SpoVNet), http://www.spovnet.de/

[2] DFG Research Group 733 "QuaP2P", http://www.quap2p.tu-darmstadt.de/

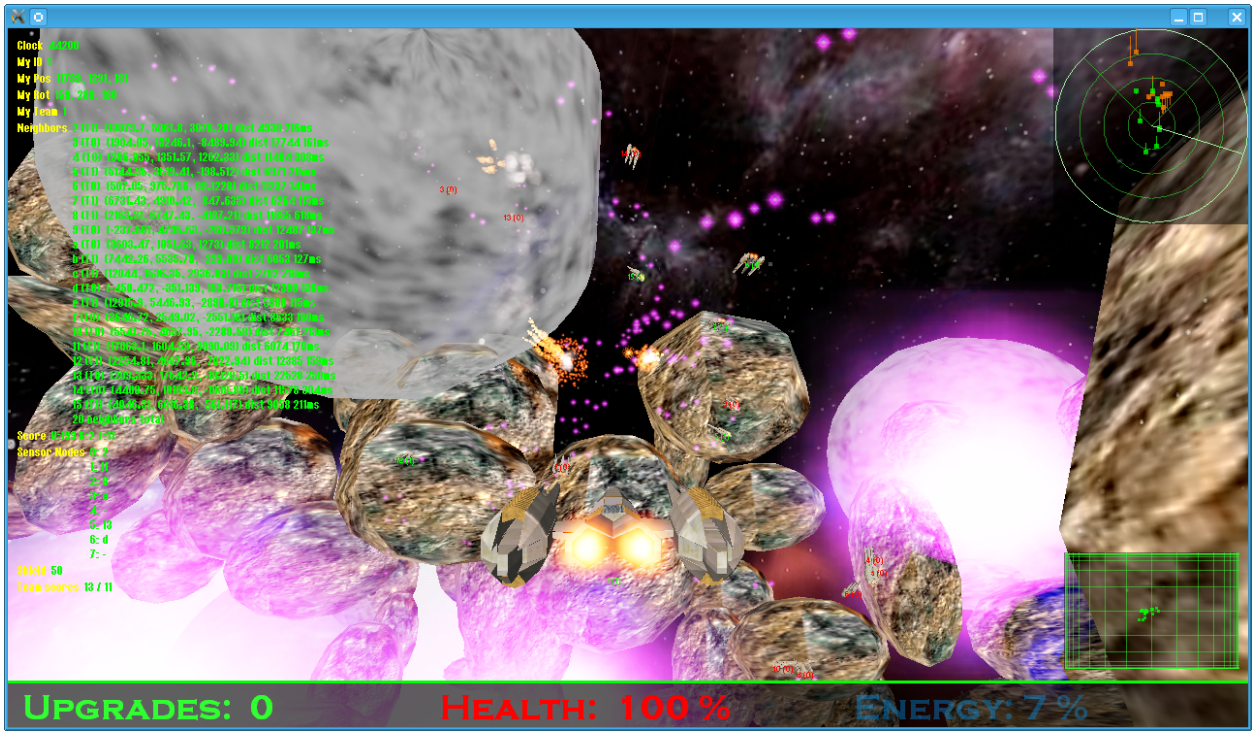


Figure 8.1.: Screen shot of the Planet PI4 game UI. Centered in the foreground is the ship controlled by the local player. The brownish bodies are solid asteroids, the gray spheres represent the regions of influence associated with a base, and the purple lights are recovery areas. In the center in front of the local player's ship a battle is going on; the purple dots are the other player's shots. The bar on the bottom indicates the local player's status information. On the top right is the radar screen with the player's surroundings, and the bottom right mini-map shows the teammates in the whole space. The top left text contains technical information provided by the game and network engine.

Figure 8.2 shows a high-level view of the software architecture. On the highest abstraction level, the software stack consists of three parts: the workload generation, which includes the Planet PI4 game logic, the application-level (overlay) network components that are to be tested, and the system and network environment, which can be embedded in a real network or as a network simulation.

8.2.1 Workload Generation

The workload generation contains the game mechanics of Planet PI4 and, as alternative workload providers, simplified mobility models that work without detailed game mechanics.

The block "PI4 Game Instance" contains the core game mechanics, including all game objects, world generation, and the 3D geometry engine using the Irrlicht engine [67]. The geometry engine provides object movement and collision detection. Irrlicht manages the 3D models and renders the UI output, if enabled. The game instance provides an interface for the local ship control (steering) and the local view of its surroundings, i.e., other ships, asteroids, etc. ("Game Control Interface" in Figure 8.2). This interface is either used by the graphical user interface presented to

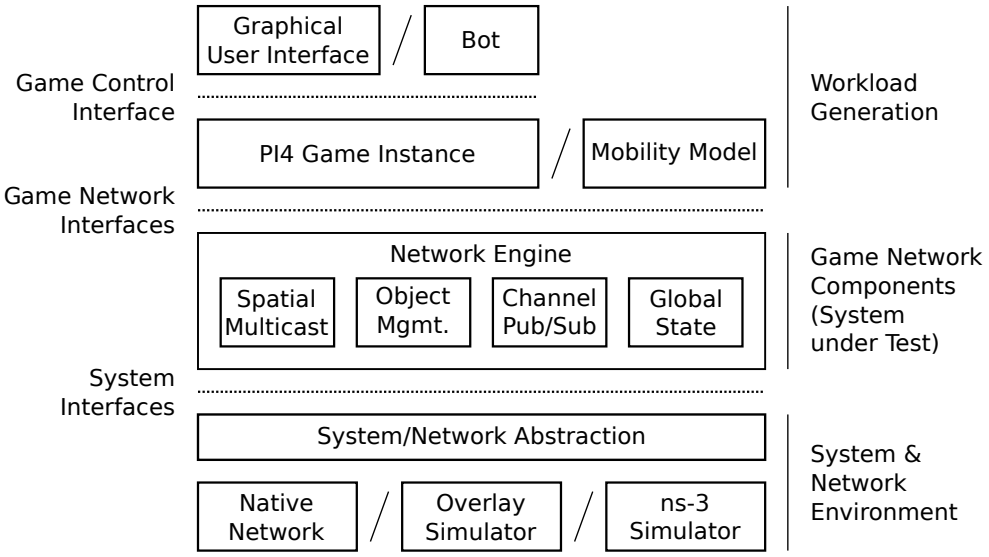


Figure 8.2.: The high-level architecture of the Planet PI4 evaluation platform. Slashes (/) indicate alternative configurations.

the human user, where the current view is rendered and mouse/keyboard input is taken from the user. Alternatively, it can be used by a bot implementation (artificial intelligence player) that plays the game autonomously. Several bot implementations are available, from very simple reactive versions to more complex ones with tactics and strategic features. These bots can be tuned to behave similarly to human players and can therefore be used to generate a realistic, interactive, and repeatable gaming workload without the need for human players [103].

Running full game mechanics together with bots has the downside of being computationally and memory intensive. This limits the number of game instances that can be run on a single machine and iterations possible within a given time interval. Furthermore, bot tuning can become complex because of their non-linear behavior [103]. For basic tests, it is therefore desirable to use a more lightweight workload generation. For this purpose, the whole game instance can be replaced with a mobility model infrastructure, which only simulates players moving in the virtual space. Different mobility patterns are supported, like random way point, random point of interest and simple swarm models. The implementation of further models is much easier than the development of a new bot. The mobility model infrastructure allows sending synthetic game update messages. Interaction like shooting each other, however, is not supported in this mode.

Both the game instance and the mobility models use the common set of game-specific network interfaces provided by the game network components.

8.2.2 Game Network Components

The network engine is split into four functional parts. Each part can be implemented independently by a separate network component, but a single component can as well provide all functions together.

Spatial Multicast is responsible for disseminating game updates among interested players. It incorporates the two aspects interest management and game event dissemination (cf. Section 4.1) and is therefore the most relevant part for this thesis.

The basic interface for spatial multicast implementations, `ISpatialMulticast`, provides the following set of functionalities:

- Getting and setting the local player ID. Player IDs are used to uniquely identify a player in the network. The selection of an appropriate ID is left to the application, which in turn can obtain an ID from the global state component (see below).
- Getting and setting the local vision range. The vision range determines the distance in the virtual space up to which local events are disseminated.
- Updating the local player position. The local player's movement is managed by the game instance. Whenever the position changes, the spatial multicast component is updated with the new position so that it can re-arrange its topology, add new neighbors, etc., whenever necessary.
- Neighbor list update notifications. The spatial multicast component manages the set of neighboring players within the vision range. This set is used by the application to display neighboring ships and allow interaction among them.
- Disseminating updates to all players in vision range. Any information that is relevant for all players in proximity should use this mechanism. Typical updates are high-frequency position updates, shooting events, or explosions.
- Sending messages to specific neighbors individually. This is useful where only a single player is to be notified, e.g., when hit by a bullet. Sending to a single player is obviously cheaper than a full dissemination.

The spatial multicast interface is typically provided by an interest management subcomponent (e.g., pSense), which can provide its own event dissemination algorithm or use InterestCast.

Object Management maintains mutable game objects. Currently, only bases are handled as mutable objects; their object state consists of the possessing team ID. The object management component may use the spacial multicast service, e.g., by handling objects as virtual participants that have their own area of interest [115].

Channel-based publish/subscribe provides a simple channel-based communication mechanism that can be used, e.g., for team chats. This is particularly used by advanced bot implementations, which exchange team orders through this channel.

Global State manages global game statistics such as player scores and player ID assignment. Since this is a low traffic component, it can most likely be implemented using a central server.

8.2.3 System and Network Environment

The system and network environment provides the necessary operating system abstraction for the application to run both in a simulated network environment and as a prototype on a real network without the need for changing the rest of the application.

The basic functionalities provided via the system interfaces are:

- To be able to run in a discrete event simulator, the application must be free of all kinds of blocking operations. All timing-related operations are therefore handled by the *task engine*. The task engine provides a simple scheduler for reoccurring and singular operations, such as updating player positions, rendering a frame, and sending update messages.

- One reason to use simulations is the determinism in execution that they provide. To achieve deterministic runs, the *random generation* used throughout the application(s) must be deterministic as well. Most network simulators provide facilities for deterministic random generation. Those are made available though the `IRandom` interface. For prototype execution, a pseudo-random generator is used.

- The *networking* functionality is abstracted as a connection-oriented datagram protocol. This abstraction allows for an exchangeability between different kinds of underlying transport protocols, such as UDP, TCP, or CUSP [155]. This interfacing layer is furthermore needed due to the fact that different network simulators provide incompatible network socket interfaces and therefore need to be translated. Connection orientation, i.e., explicit connection initiation and teardown, is necessary to support connection-oriented protocols like TCP and CUSP. Since the application protocols work with message packets instead of data streams and to support datagram protocols (UDP) as the underlying transport, the interface uses the datagram abstraction.

In addition, runtime-independent instrumentation for the testing and evaluation of the prototype algorithms is facilitated through a set of additional common interfaces:

- A simple *logging* facility to output per-module log messages with different log levels. Using the C++ logging framework log4cplus[3], log messages can be output on the console during runtime, forwarded to the respective simulator logging facility, or written directly to an SQLite database. Persistent recording of log messages can be controlled based on module name and log level to save storage space and execution time.

- The *statistics* facility is used for most measurement purposes. It is useful wherever numeric metrics are to be recorded for an evaluation in the postprocessing phase. The statistics data model is based on a previous testbed design [106]. Each statistic is identified with a name and has additional meta-data like a unit and a human-readable label. Numeric values can either be pushed or be pulled in regular intervals, typically one second. A statistic can either be associated with a node, meaning that its measurement data stems from that node, or it can be global, typically meaning that it contains an aggregate over several nodes. Statistics can build

[3] log4cplus, Logging Framework for C++, http://sourceforge.net/projects/log4cplus/

hierarchies, i.e., a statistic can have parents to which it propagates its measurement value in each recording interval. A typical example is the instantiation of one statistic per node for a given metric (e.g., number of neighbors), which all have a common parent statistic that stores the global aggregates. Individual statistics can be set to persistent or non-persistent, controlling whether their data is stored to the experiment database. Non-persistent statistics can be used to just forward aggregated measurements to their parent(s). For each measurement interval, each statistic stores an aggregate of the accumulated samples consisting of the number of samples, their minimum and maximum value, their sum and sum of squares. This way, the data amount is minimized while keeping the necessary information for statistic evaluation (mean, variance) and further aggregation. Alternatively, where needed, the full set of individual samples can be stored for more detailed analysis (e.g., the calculation of quantiles) during postprocessing.

- For cases where the monitored data is structured in a way that it cannot be broken down into statistics of numerical values (e.g., tables of related columns), or the data is simply non-numeric, *state dumps* are an alternative way of persisting system state for postprocessing. Like the statistics, dumps are identified by a name. Each dump regularly, typically once per second, pulls its data in the form of a string from each node. The data can for example be comma- or tab-separated lists of entries per node. Together with a header line stored as the prefix in the dump metadata, the data from multiple nodes can later be joined and processed as a combined structure. Since dumps are pulled synchronously in discrete-event simulations, they provide a consistent view on the system state over all nodes.

Logging, statistics, and dump data can be directly written to an SQLite database or passed to the corresponding simulator facilities, if available.

8.3 Experiment Workflow and Data Model

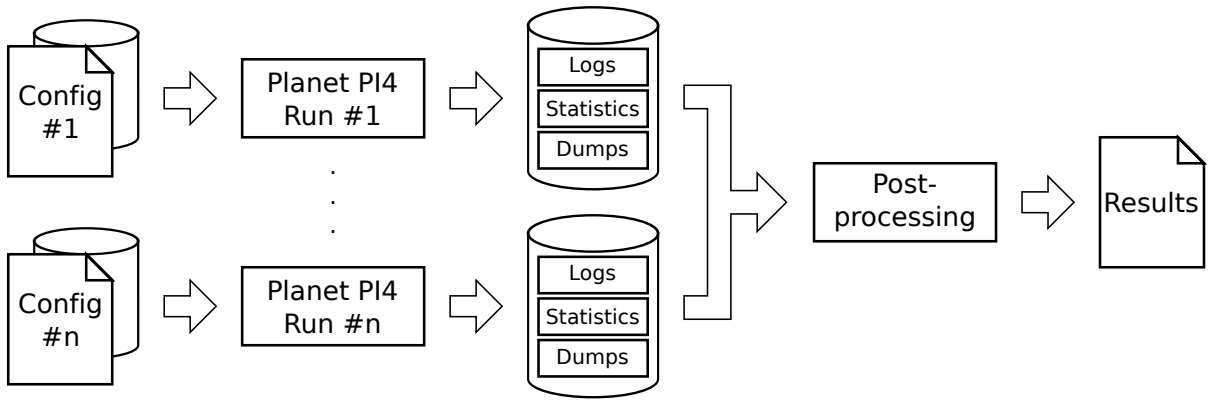


Figure 8.3.: The basic experimentation workflow. For each run, a configuration is specified for both the Planet PI4 and network simulator settings. Each run generates a database of logs, statistics, and dumps, which are evaluated in the postprocessing phase.

Figure 8.3 shows an overview of the experimentation workflow. The configuration for each run consists of the Planet PI4 game configuration and the network simulator configuration, depending on whether and which simulation is used. Planet PI4's configuration consists of a set of configuration variables, which can be specified via a configuration file and the command line. The configuration variables are available to all components throughout the application. This configuration determines the selected network components and workload model, among others. In addition, there is the network simulator configuration, which determines the random seed, network topology, number of nodes, bandwidths, etc.

During each run, either in a simulation or in a real network, the Planet PI4 framework allows all its components to write log messages, statistics, and dumps. This data is stored in a common SQLite database for each experiment run. The outputs from multiple runs are integrated in the postprocessing phase, comparing different configurations and for merging results of experiments with different random seeds for statistical significance.

9 Evaluation

In this chapter we present the evaluation scenarios and results of the evaluation of InterestCast. First, we elaborate on the main questions this evaluation aims to answer. For that, we start with the more general and high-level questions and consequently move towards the details specific to InterestCast. The high-level questions can be answered using macro metrics and higher network abstractions, while the detail questions are answered using micro metrics and the more fine-grained network models.

For all evaluations we use settings based on gaming or virtual environment scenarios, using InterestCast as the spatial multicast solution. Since InterestCast purely relies on information from the node's local neighborhood, the total network size in terms of the number of nodes is not a critical factor. On the contrary, the density of the network, i.e., the number of participants per area in the virtual space, and therefore the density (as a graph metric) of the interest graph plays a significant role, as it has a direct effect on the average number of neighbors in the interest graph. Hence, we can use moderate numbers of participants (typically between 50 and 200) to achieve the desired variations in density.

The evaluation aims for answering the following main questions:

How can the utility-based local optimization balance load and performance? This first question targets the general effects of the local optimization using utility functions. Initially, we want to demonstrate the behavior of the optimization process. Further, we analyze the optimization results depending on the network size, clustering, and with different utility function weights.

The analysis of these basic properties is conducted using a graph-based routing model, a workload based on interest graphs, and a simple distance-based network model. Here, the interest graph is synthetically generated using a vision-range based virtual reality model. The network model includes end-to-end latencies between nodes and a static per-node bandwidth model. Each message flow allocates a fixed amount of the nodes' bandwidths. Hence, the model abstracts from many network details, such as network dynamics and overload effects such as queuing. This high abstraction allows for an analysis concentrating on the local optimization without the need for dealing with many side effects.

The graph-based model and the basic optimization behavior are presented in Section 9.3.

How can the trade-off between bandwidth and latency be tuned? The utility weights (cf. Section 5.5) provide the easiest tuning knob for the trade-off between bandwidth demand and achieved latency. The effect of changing their ratio is demonstrated in Section 9.3.5.

What is the effect of interest clustering on InterestCast's optimization capabilities? We expect InterestCast to profit from neighbors with common interests due to its forwarding and aggregation concepts. This question aims for evaluating this potential. Here, we use the

same model and simulation as above, but with a different interest graph. To gain graphs with specific clustering properties, the interest graph is built using a synthetic random graph generator.

How close to the global optimum are the solutions of the local optimization algorithm? To answer this question, we use the integer programs described in Section 5.6 and compare their results with those of the local optimization algorithm. Again, we use the highly abstracted network model, as the integer programs provide offline solutions restricted to the routing configuration. They do not consider message scheduling, queuing, and other network dynamics at runtime. This way, we can most directly compare the solutions of the two approaches. Despite the simplifications in the models, however, the main network properties are considered, allowing us to view InterestCast's results in relation to the global optima.

How does InterestCast perform in comparison with purely direct P2P connections? This question is about the net performance of InterestCast compared to the alternative approach of simple direct connections within the interest set, as used by gaming overlays like pSense [143] and VON [78]. To analyze the net performance, we have to use the full InterestCast stack. We therefore use the prototype implementation described in Chapter 7 and the Planet PI4 testbed (Chapter 8). We compare the plain pSense implementation from Planet PI4 with the combination of pSense as the interest management component and InterestCast as the event dissemination component. We use the overlay simulation mode of Planet PI4 to obtain realistic results for gaming/NVE scenarios in fixed networks.

What is the overhead of the node state exchange necessary for local optimization? As discussed in Section 6.2, InterestCast's optimization process relies on measurement data from the nodes' neighborhoods and the paths they are involved in. This question is about the effective communication overhead caused by InterestCast's state exchange protocol (cf. Section 6.3). We measure this amount with the same simulation setup as above, but using micro metrics differentiating the traffic generated by InterestCast's message types.

To answer these questions, we first give an overview of the evaluation modes. We use a two-staged model, a simple graph-based version for the general behavior and a detailed application and network simulation for the full protocol evaluation. Then, we present the important factors and metrics for the evaluation. Afterwards we first show the results of the graph-based model evaluation, followed by the detailed protocol evaluation. Finally, we discuss the results.

9.1 Evaluation Modes

To answer the above questions, we need different ways for solving the InterestCast problem. Accordingly, we use three distinct setups:

1. Global solutions using a standard integer programming solver. Here we directly use the integer programs for the global optimization problem, as derived in Section 5.6 and provided in Appendix A. They are solved using the SCIP solver [64]. The integer problems consider a

1. Global solution	2. Graph-based local optimization	3. Full protocol implementation
Integer programming	Custom graph model	InterestCast protocol
SCIP + Zimpl	Scala	C++
Global optimum	InterestCast's incremental optimization	
	Interest graph	Virtual reality scenario
	No mobility	Various mobility models
	Simple delay/bandwidth model	Network simulator or real network
	Static event streams	Gaming workload

Table 9.1.: Overview comparing the three evaluation modes

static scenario, which can be considered as the configuration at a given point in time. The optimization finds the optimum for that configuration without considering any dynamics such as transition costs. For being able to represent the problem as integer programs, we use the simplified workload and network model. The interest graph is directly provided as an input, as well as the node bandwidths. Inter-node latencies are calculated on a two-dimensional geographical distance model.

2. Custom graph-based simulation. This mode runs InterestCast's incremental local optimization algorithm in a round-based simulation. The network and workload models are compatible with those used for the global optimization. This way, their results can be compared so that the global optimization serves as a benchmark for InterestCast's algorithm. Further, the simplified models allow analyzing the basic behavior without effects induced by network dynamics and interdependencies. The detailed simulation setup is described below (Section 9.3).

3. The full prototype implementation. Finally, using the full protocol implementation as described in Chapter 7, we evaluate the system with network dynamics and all necessary overhead. This provides the proof of concept from the systems perspective, i.e., that InterestCast can run as a real distributed system. Embedded in the Planet PI4 evaluation environment (Chapter 8), network simulators provide a realistic and reproducible network environment as well as various workload scenarios.

Table 9.1 provides an overview and comparison of the three modes. Accordingly, we use two stages with distinct models for the interest graph and the underlying network model. The first (Section 9.3) is commonly used by the first two evaluation modes, making their solutions comparable. The second (Section 9.4) provides a proof of concept for the full protocol implementation.

9.2 Important Factors and Metrics

For the evaluation, it is important to consider an appropriate set of factors (i.e., relevant input parameters) and metrics. This section provides an overview of the factors and metrics to be used in the following.

9.2.1 Factors

Number of nodes. The number of nodes, or participants, is an important factor for almost every distributed system. The state managed by InterestCast, however, does not directly depend on the total number of nodes, but rather on the number of participants of interest, i.e., interest set size.

Interest set size. In a virtual reality scenario, the interest set size can depend on the total number of participants, but also on the virtual world size and the participant distribution in the virtual world. With a uniform distribution and a fixed-size virtual world, the average interest set size scales about linearly with the number of participants and can therefore be controlled via the latter. We will use this method in the following.

The interest set size must further be considered in relation to the event throughput and node capacities. The product of interest set size and event throughput gives the net bandwidth demand for direct connections, which is a critical factor. We keep the event throughput per participant constant while varying the interest set size.

Clustering. InterestCast's event forwarding preferably chooses forwarders who are also interested in the respective events to achieve maximum savings. The likelihood for finding such forwarders depends on the clustering of the interest graph. A higher clustering coefficient increases the number of potential forwarders with interest overlaps and therefore increases the optimization potential.

Mobility. We consider mobility with respect to the virtual world. This mobility determines the dynamics of the interest set as well as interest levels. An important aspect of the mobility is the velocity of participants in the virtual world. In addition, the mobility patterns play a role, such as the movements of participants with respect to each other. A completely independent movement, such as that generated by simple uncoordinated random waypoint models, has the fastest relative velocities and thus a high interest set change rate. Uncoordinated models therefore provide the most challenging workload, making pessimistic assumptions with respect to the mobility in real applications. A more coordinated and more realistic workload is provided by using bots and the full Planet PI4 gameplay.

9.2.2 Metrics

Path latency. Considering latency-sensitive applications, the event delivery latency is one of the primary metrics. Path latency refers to the accumulated latency incurred by the event delivery

over possibly multiple overlay hops, including queuing and network propagation delays, as defined in Equation 5.1.

Since the latency can be measured for each individual event that is delivered, the latency distribution is of interest. For a comparison, we have to use aggregate values. Important aggregates are the mean value, maxima, and quantiles.

Path latency stretch. While the absolute latency is relevant for the application, it is highly dependent on the latencies of the underlying network. The path latency stretch serves as a more network-agnostic metric by taking the ratio of effective path latency and the network latency for a direct connection between source and destination. Hence, for the direct dissemination case, the latency stretch is always one, if we ignore queuing delays.

Missing neighbors. In extreme overload situations, connecting to neighbors may fail entirely. To be aware of this case, we also consider missing neighbor ratio.[1]

Staleness. For events updating continuous state, staleness is the most relevant metric. In particular, staleness accounts for the loss of updates, in addition to the update rate. Staleness is quantified as the average age of a piece of information, e.g., a position, from a neighboring player.

Bandwidth demand. With bandwidth demand being the second optimization factor, the gross bandwidth demand of the participants is an important metric as well. Depending on the simulation model, the gross bandwidth demand includes all additional communication needs, such as monitoring and coordination overhead. We consider packet header overhead down to Layer 3 (IP), but ignore any protocols below this layer. This avoids making assumptions on the network technology on the lowest layers.

Like for latency, the bandwidth distribution among the nodes is of interest. Likewise, we consider mean, maximum, and quantile aggregates for the analysis.

Total utility. Finally, the total utility, i.e., the optimization objective (cf. Equation 5.20), is a core metric for comparing different solutions.

9.3 Graph-Based Model

The basic properties of InterestCast's incremental optimization algorithm are evaluated using graph based models for interest and routing, allowing us to focus on the algorithm's behavior. The interest graph model allows using interest graphs (cf. Section 5.1) directly, generated either based on a concrete scenario or synthetically with desired properties. For the scenarios analyzed with this model, we assume static interest graphs and static network properties. Results from the graph-based model have been published earlier [101].

[1] The interest set selection can be interpreted as a classification problem in which all nodes have to be classified either as in the interest set or as of no interest. The missing neighbor ratio is then the inverse of the precision metric for this classification problem.

In each round of the round-based simulation, each node performs one optimization step of the algorithm. The round-based model abstracts from the frequency of iterations, which can be adjusted to the application needs. A synchronization in rounds is in fact not needed for the algorithm, but is used here for the simplicity of the evaluation.

The network and host model used for the simulations is based on abstracted geographical locations adapted from the Global Network Positioning (GNP) [124] model. Unlike the GNP model, however, we use uniform node positioning. This is a pessimistic assumption compared to the more clustered real-world global host distribution. Host upstream bandwidths are modeled in multiples of the bandwidth required for a single event stream, while all event streams are assumed to have equal bandwidth. We use a uniform upstream bandwidth distribution between 20 and 100 outgoing streams. This corresponds to node uplink capacities ranging from 100 to 500 kbit/s, Assuming a stream throughput of 512 bytes/s. This corresponds to, e.g., 10 messages/s of 50 bytes each. This bandwidth distribution, again, is rather pessimistic, underestimating the heterogeneity of today's typical Internet uplink capacities, which range from 100 kbit/s to 10 Mbit/s. The local link latency adds 10 ms to each hop. Further details, such as jitter, node churn, mobility, and firewalls, are omitted in this model.

We assume messages to have an overhead from packet headers in the same order of magnitude as the payload. For small messages, this assumption is realistic: using UDP transport, IPv4 and UDP headers have a total size of $20 + 8 = 28$ bytes. With TCP, this grows to $20 + 20 = 40$ bytes, and IPv6 makes it even worse (cf. Section 2.1).

9.3.1 Virtual Reality Scenario

In the virtual reality scenario, a variable number of nodes is placed randomly in a fixed-size region of 1000×1000 units. The interest function is a Gaussian of the Euclidean virtual world distance.

$$I(v,u) := e^{-\frac{\|v,u\|^2}{2c^2}}, \quad c := r\left(2\,ln\frac{1}{\tau}\right)^{-0.5}.$$

c is calculated so that for a given threshold τ, $I(v,u)$ comes below τ as the distance of v and u exceeds r. We choose $\tau = 0.1$ and a default vision range of $r = 250$. This function is more complex than a linear interest level decay with distance, but reflects the needs of a typical virtual reality application better: Up to a certain distance, neighbors are well visible and thus of high interest. More distant neighbors quickly become less interesting, and at the border of the vision range, i.e., at the tail of the Gaussian, they are barely visible and thus do not need to be presented with high precision. At the end of the interest range, on the other hand, In addition, it increases the number of high-interest neighbors compared to a simple linear function and thus leads to a more uniform distribution of neighbors across interest levels.

As utility functions, we choose suitable candidates for the virtual reality scenario from the functions discussed in Section 6.5:

$$\mathcal{U}_{\text{lat}}(\mathcal{L},\iota) := 1 - \left(\frac{\mathcal{L}}{0.1 + 0.9 * (1-\iota)}\right)^2 \quad \text{and} \quad \mathcal{U}_{\text{bw}}(b) := 1 - \max(4 \cdot (b - 0.5), 0)^3$$

Nodes	Avg. interest set size	Clustering coefficient
50	7.64	0.642
100	14.93	0.637
150	23.90	0.645
200	30.57	0.642

Table 9.2.: Properties of the synthetic virtual environment interest graph

The latency utility function $\mathcal{U}_{\text{lat}}$ has a quadratic decrease with the latency. Further, it it normalized with respect to a latency of 0.1 s for highest interest and 0.9 s for lowest interest. The bandwidth demand utility function maps to a value of one up to a relative bandwidth demand of 0.5 and then decreases cubically. The utility weights are set to

$$w_{\text{lat}} := 1 \quad \text{and} \quad w_{\text{bw}} := 10$$

to compensate the on average higher total number of subscriptions than nodes and therefore the otherwise higher weight of $\mathcal{U}_{\text{lat}}$ in the sum of all utilities.

Basic Virtual Reality Interest Properties

Table 9.2 details important properties of the interest graph of the given virtual reality scenario. Keeping the world size fixed and varying the number of nodes between 50 and 200, the average fan-out ranges from 7 to 31. Since the algorithm works on each node's local view, the total number of nodes does not play a significant role. The relevant factor is the fan-out, i.e, the interest set size. Intuitively, with a fixed virtual world size, the fan-out grows linearly with the node density. The clustering coefficient lies around 0.64, independent of density.

9.3.2 The Basic Optimization Process

The iteration process in an exemplary static scenario is illustrated in Figures 9.1 and 9.2, showing the bandwidth demand, path length, and latencies after each iteration. Nodes do not change positions, capacities, or interest between the iterations. Latency stretch, interest-latency stretch, and path length are normalized with respect to the initial direct dissemination configuration. Interest-latency stretch is the average product of latency and interest level of all paths, therefore laying a higher weight on high-interest paths. This metric shows that high-interest paths are penalized significantly less than the average path. Figure 9.2 shows the distributions of relative bandwidth demand and path latency stretch. The average bandwidth demand is reduced from almost 100% to less than 75%. Initially, (line "0" in Figure 9.2), more than 35% of the nodes have a relative bandwidth demand of greater than 100%, i.e., are overloaded. After 31 iterations, (line "31"), the ratio is reduced to less than 3%. The plot illustrates the load shift: Overloaded nodes are relieved (Figure 9.2a, top right) at the expense of sparsely utilized nodes (bottom left). Eventually, the majority has a relative bandwidth demand between 60 and 80%.

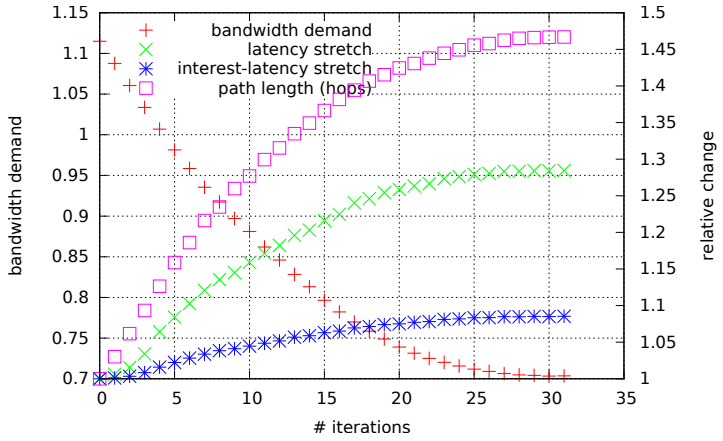
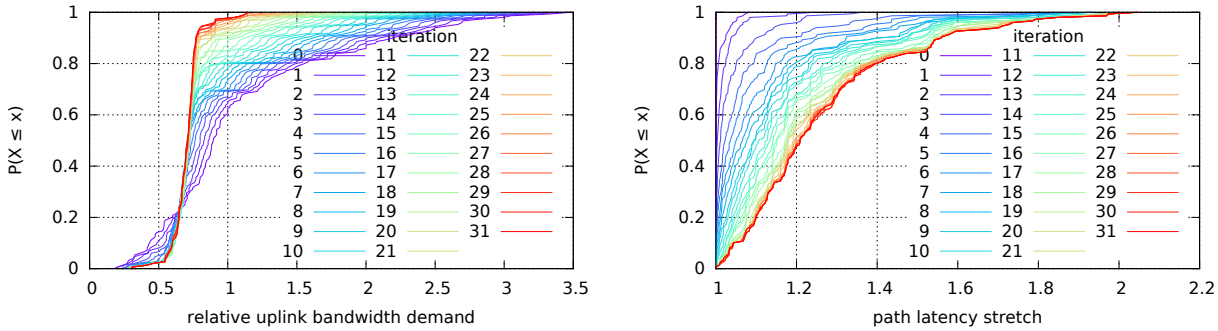


Figure 9.1.: Iterations of an InterestCast optimization process with 150 nodes.



(a) CDF of relative bandwidth demand

(b) CDF of path latency stretch ($\frac{\mathscr{L}}{\ell_{\text{net}}}$)

Figure 9.2.: Iterations of an optimization process with 150 nodes

9.3.3 Comparison with Global Optimization

In this section, we compare global solutions to the optimization problem with the results of the InterestCast algorithm. For the comparison, we use the graph-based model described in Section 9.3. We vary the total number of nodes, with it the node density in the virtual reality scenario, and thus the average node degree in the interest graph. Further, we stick to the static configuration, as the global optimizer only provides solutions for one particular point in time.

We concentrate on two main aspects:

1. the time needed by the global optimizer to compute the solution (solve time), and
2. the comparison of the resulting solutions of the two approaches, especially with respect to the global utilities.

For this test, we use the same virtual scenario settings as above, except that the vision range is increased to 500 units. This is done to achieve a sufficiently high node density despite the necessarily lower number of nodes due to the problem size limitations of the solver (see below). The problem size is varied from 10 to 50 nodes. For each problem size, five scenarios are randomly generated.

SCIP	3.1.0	gcc	4.8.2
SoPlex	2.0.0	Zimpl	3.3.2
Ipopt	3.11.7	CppAD	20140000.1
GMP	5.1.3	zlib	1.2.8

Table 9.3.: Versions of the software components used with the SCIP solver

We test both integer programming variants defined in Section 5.6 and detailed in Appendix A:

ILP The simplified linear version, which minimizes latency while keeping bandwidth bounds (cf. Section 5.6.1).

MINLP The nonlinear version using utility functions and therefore the actual InterestCast optimization problem (cf. Section 5.6.2).

The global solutions are computed using the SCIP solver [64] in version 3.1.0 on a 64-bit Linux machine. SCIP is considered one of the fastest non-commercial solvers for mixed integer programming (MIP) and mixed integer nonlinear programming (MINLP).[2] The software version details used for this evaluation are given in Table 9.3.

The solver is configured to stop at a relative gap between primal and dual problem[3] of 1% or less to speed up the optimization process while maintaining reasonable errors. Further, the maximum solve time is limited to 24 hours (86,400 s). These settings can lead to suboptimal solutions. As long as the time limit is not reached, however, the relative error of 1% is acceptable for the comparison. Solutions taking more than 24 hours to compute seem infeasible for any practical application. Suboptimal solutions can have inconsistent routes because duplicate routes for a single subscription are not disallowed by the constraints of the given integer problems for reasons of simplicity. Optimal solutions, however, usually do not contain such duplicates due to their unnecessary costs. We therefore ignore those routes in the evaluation.

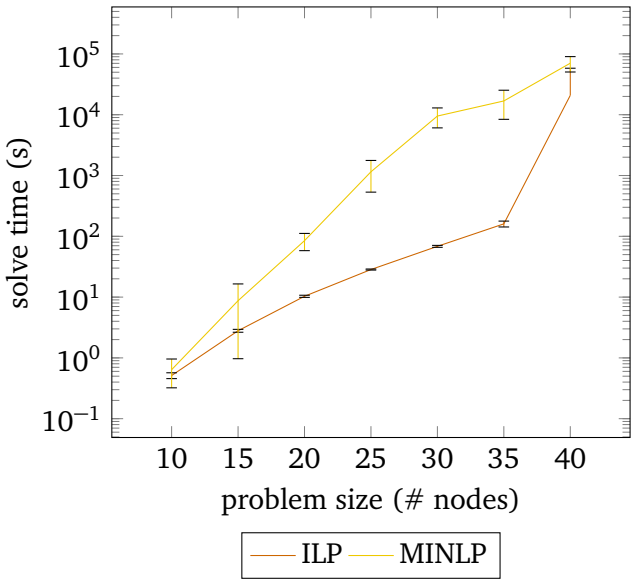
Solve Time

Figure 9.3 shows the solve times for the ILP and MINLP problems depending on the problem size. The values represent the averages of the five runs per problem size. The numbers show an exponential growth in solve time. The linear ILP problem is solved significantly faster but its solve time also grows quickly with the problem size. Within 24 hours, not even all problem instances with only 40 nodes were solvable to the 1% gap. The solve times further show a high variance as indicated by the error bars in Figure 9.3.

Hence, even with moderate problem sizes, the optimal (or even close to optimal) global solutions take infeasible amounts of time to be computed on demand. InterestCast's incremental

[2] According to the MIPLIB2010 [63] benchmark results by H. Mittelmann (http://plato.asu.edu/ftp/milpc.html, accessed 2015-06-03), graphically illustrated on http://scip.zib.de/ (accessed 2015-06-03). According to the benchmark results, commercial products are up to one order of magnitude faster, which, however, is not relevant for the purpose of comparing the results.

[3] The relative gap between primal and dual problem is computed as $\text{gap} = \frac{|\text{primal}-\text{dual}|}{\min(|\text{dual}|,|\text{primal}|}$ (http://scip.zib.de/doc-3.0.1/html/PARAMETERS.shtml, accessed: 2015-06-09) and provides an estimation on the distance to the lower bound of the solution for the optimization problem.



Size	ILP	MINLP
10	0.5 s	0.6 s
15	2.8 s	8.7 s
20	10.3 s	84.8 s
25	28.4 s	1151.6 s
30	68.4 s	9506.4 s
35	160.2 s	16845.4 s
40	20812.4 s	70396.2 s

Figure 9.3.: Integer programming solution times depending on the problem size (number of nodes) on an Intel Xeon E5-2650 at 2.0 GHz

optimization algorithm on the other hand only requires a small number of iterations, as shown in Section 9.3.2.

Comparison of the Results

InterestCast's faster optimization comes with the trade-off of a suboptimal solution. To find out about the degree of suboptimality, we compare the solutions computed by the integer programming solver (ILP and MINLP) with the results from the InterestCast incremental optimization algorithm. For this purpose, the algorithm is run with the same static configurations as given as input to the integer programming solver. The iterations are run until a convergence is reached, as visualized in Figure 9.1. In addition, we provide the metrics of the pure direct communication mode, where each node sends its updates to all receivers individually.

Figure 9.4 plots the latency stretch ($\frac{\mathscr{L}}{\ell_{\text{net}}}$) over the relative bandwidth demand for each individual run. Hence, there are five bubbles for each problem size and each optimization mode. For the direct communication, the latency stretch is always one. The bandwidth demand, however, is the highest. The ILP mode uses message forwarding only to keep bandwidth within capacity limits (i.e., a relative bandwidth demand of at most one for each node). The latency stretch is therefore minimal. In contrast, the utility function based MINLP mode reduces bandwidth demand much further, at the cost of a higher latency stretch. For the small scenarios considered here, the added latency is still very low with less than 5%. The InterestCast optimization reduces the bandwidth demand to about the same level, but with an even higher latency stretch of about up to 12%.

Figure 9.5 shows more aggregated results. Here, one can see that the absolute impact of the higher latency stretch is rather marginal (Figure 9.5c). The total utility according to the utility functions is shown in Figure 9.5d. The total utility is massively increased by all of the optimization approaches compared to the direct communication case, mostly due to the bandwidth demand reduction (cf. Figures 9.5a and 9.5b). With respect to bandwidth demand, the difference between

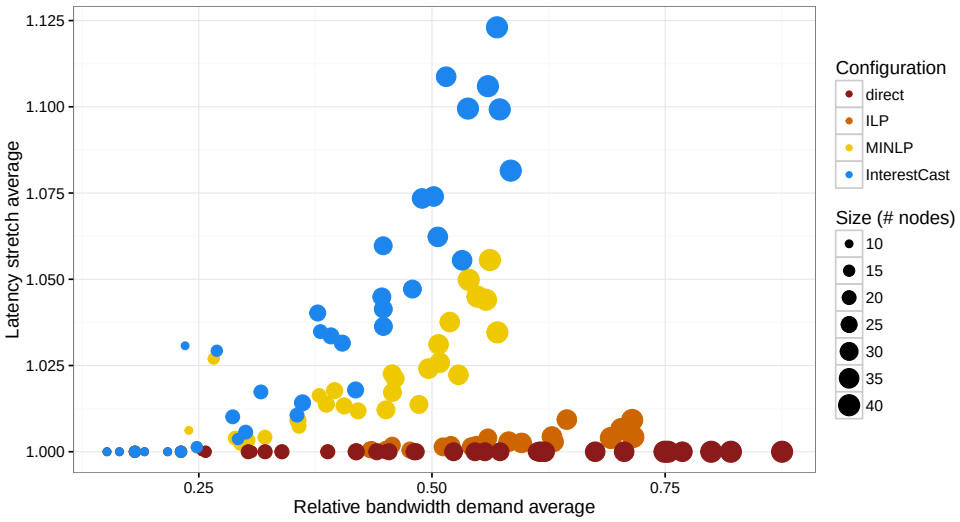
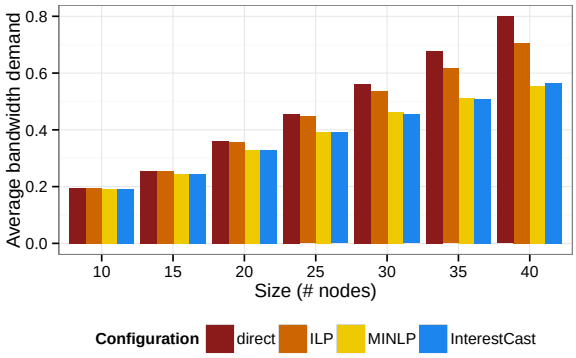
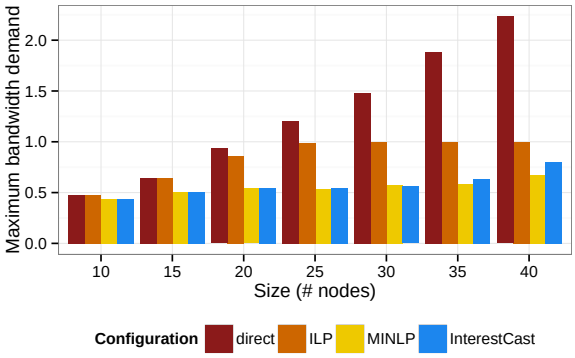


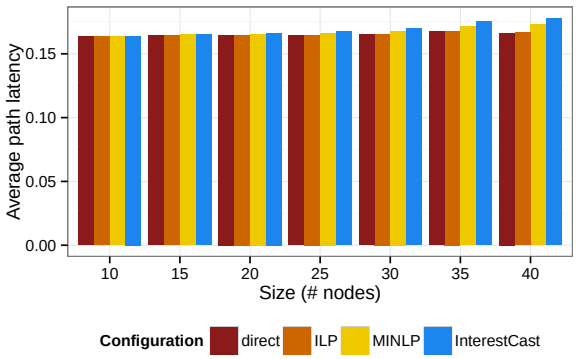
Figure 9.4.: Latency stretch over relative bandwidth demand for unoptimized direct communication, ILP solution, MINLP solution, and the result of the InterestCast incremental optimization algorithm. The bubbles show the individual results of the five runs for each problem size, which is encoded as the bubble size.



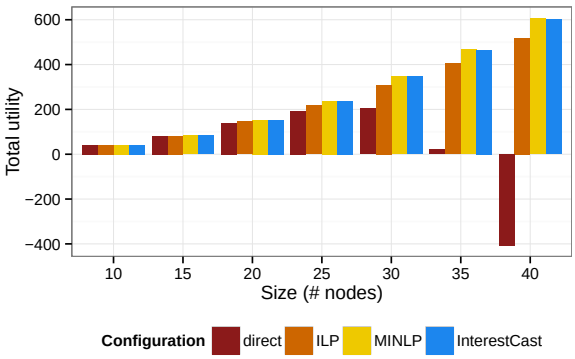
(a) Average bandwidth demand by problem size



(b) Maximum bandwidth demand by problem size

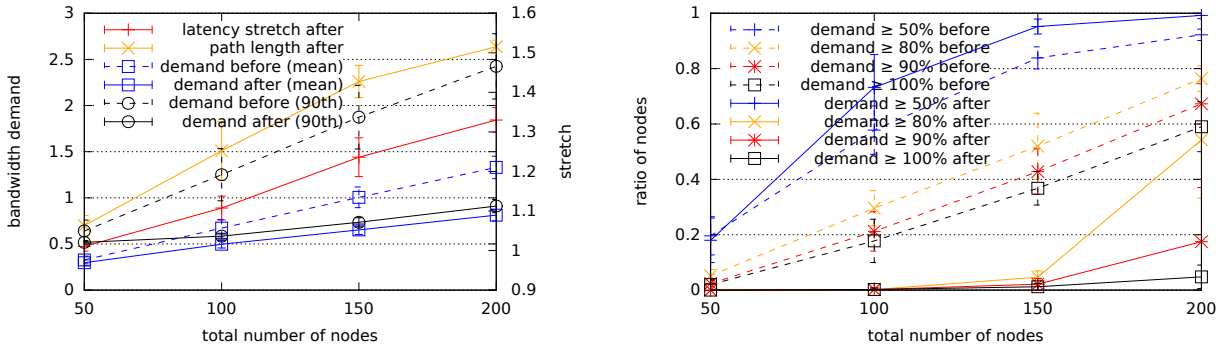


(c) Path latency by problem size



(d) Total utility by problem size

Figure 9.5.: Comparison of the three optimization approaches ILP, MINLP, and InterestCast as well as the unoptimized direct communication.



(a) Relative bandwidth demand before (i.e., direct communication) and after the optimization

(b) Ratios of nodes exceeding a certain relative bandwidth demand

Figure 9.6.: Bandwidth demand by varying number of nodes

the ILP and MINLP is notable. The ILP problem only considers the bandwidth cap and therefore maxes out the available bandwidth (9.5b), while the utility-based MINLP considers the bandwidth utility function and thus keeps the demand lower at the cost of some more latency stretch, as does InterestCast.

The main result, however, is that InterestCast's optimization can almost keep up with the global integer programming solution. Hence, the incremental local optimization, which is targeted for a completely distributed setting, achieves good results compared to the global optima, in particular regarding the significantly lower solve time.

9.3.4 Node Density

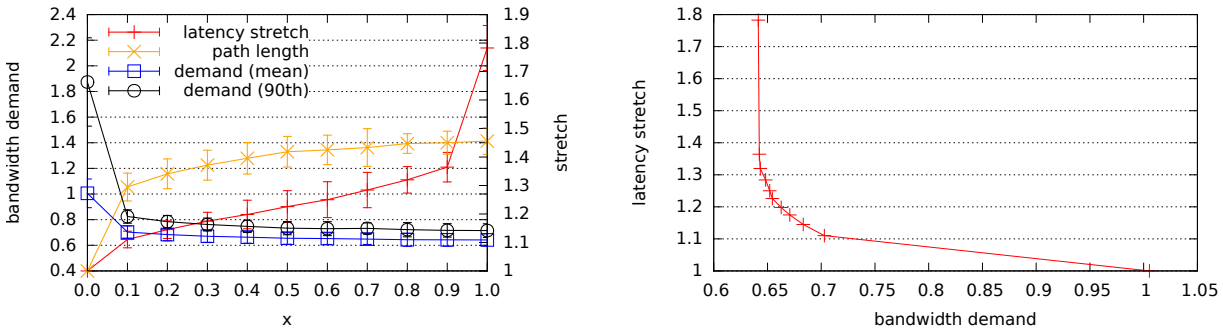
In the following, we only look at InterestCast's optimization properties. We now scale the number of nodes beyond what is solvable using integer programming.

Figure 9.6a shows the initial and optimized uplink utilizations as well as path length and latency stretch depending on the number of nodes. Both mean and 90th percentile of the initial relative bandwidth demand grow linearly with the number of nodes, because the number of outgoing message streams equals the fan-out (cf. Table 9.2). With 50 nodes, bandwidth demand is below 100%, hence, there is no need for further reduction. But the higher the density and therefore bandwidth demand gets, the greater becomes the improvement in the optimized case. Especially the most utilized nodes, represented by the 90th percentiles, profit the most. In return, reduction of bandwidth demand in the denser cases is paid with an increased path length and latency stretch.

For the same scenarios, Figure 9.6b shows the ratios of nodes with a bandwidth demand over 50, 80, 90, and 100% respectively, before and after optimization.

9.3.5 Utility Weights

To demonstrate the trade-off between latency and bandwidth demand, Figure 9.7a shows the results for 11 weight settings between "free" bandwidth (0.0 on the x-axis) and "free" latency (1.0). The values at 0.0 represent the purely direct delivery configuration, since only latency has



(a) Path length, latency stretch, and average bandwidth demand by weight factor

(b) Latency stretch by average bandwidth demand, visualizing the trade-off

Figure 9.7.: Relative bandwidth demand and path lengths depending on weight factors of the utility function. $w_{\text{bw}} = 10 \cdot x$, $w_{\text{lat}} = 1 - x$.

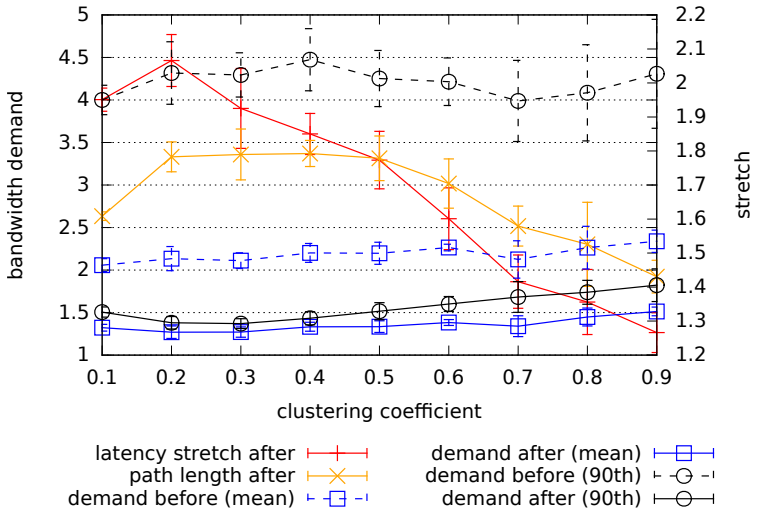


Figure 9.8.: Optimization potential depending on clustering coefficient

costs associated and the used delay model never violates the triangle inequality. At the other end, when latency does not have any costs associated, the last bit of bandwidth saving is bought with significantly added latency.

Between the extreme cases, the weights can be used to adjust the preference between bandwidth demand and indirection latency. Figure 9.7b visualizes this trade-off by plotting the latency stretch over the respective bandwidth demand for the data points of Figure 9.7a.

9.3.6 Clustering

To examine the optimization potential depending on the clustering, we use a synthetic graph generator to build connected subscription graphs. A variant of the random intersection graph algorithm by Deijfen and Kets [50] allows us to generate random graphs with a predetermined degree distribution and clustering coefficient. For the node degree generation, we use a Gaussian distribution. The algorithm's parameters are tuned to generate graphs with a Gaussian distributed node degree

Class	Name	Connection	Downlink (bytes/s)	Uplink (bytes/s)
Strong	player25MBit	25 MBit/s VDSL	3200000	640000
Medium	player6MBit	6 MBit/s ADSL	768000	64000
Weak	player1MBit	1 MBit/s ADSL	128000	16000

Table 9.4.: Node connection classes used for the network simulation

with an average of 25 and clustering coefficients of $0.1, 0.2, ..., 0.9$. The nodes' link capacities are reduced so that the initial mean relative bandwidth demand is above 2 and the 90^{th} percentile is above 4.

The results (Figure 9.8) show that a higher clustering coefficient increases the potential for optimization. Surprisingly, this is barely expressed by a decreasing bandwidth demand after the optimization. Instead, however, a similar bandwidth demand can be achieved with a significantly lower latency stretch for higher clustering coefficients. The behavior of the algorithm is a result of the utility function used. The utility function can therefore also be tuned according to application needs. Overall, the results show that a high clustering improves the performance of our algorithm.

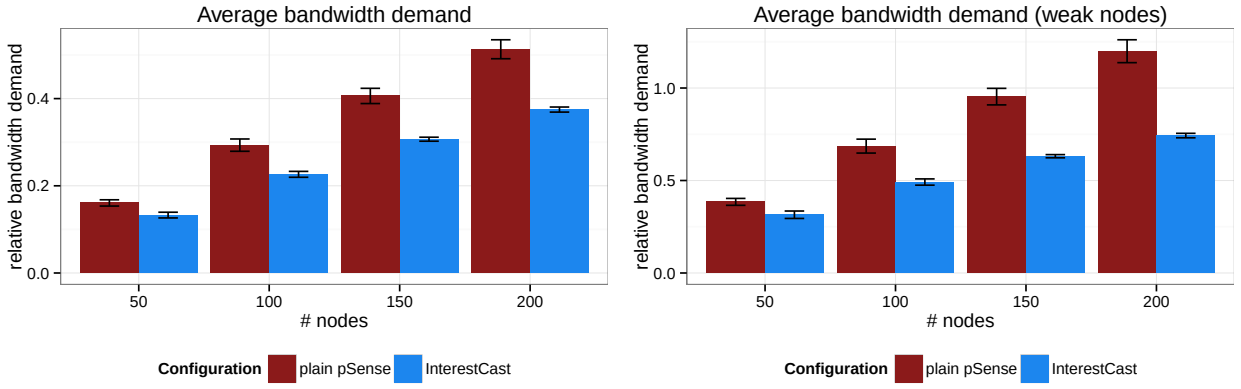
9.4 Prototype

The experiments in the following are conducted in the third evaluation mode defined in Section 9.1, i.e., with the InterestCast prototype implementation (cf. Chapter 7) running in the Planet PI4 framework (Chapter 8) on a network simulation. The full protocol implementation shows how the eventual system performs in real-world settings.

The baseline scenario settings are chosen analogously to those of the virtual reality scenario for the graph based model (Section 9.3.1). An implementation of pSense [143] is used for the interest management on top of InterestCast. The focus of the evaluation in this mode is on the comparison with plain pSense using direct dissemination and on the InterestCast protocol overhead.

If not indicated otherwise, the following network settings are used. The emulated network environment is provided by the packet-level overlay simulator described in [106], which was previously used for large-scale overlay network simulations [104]. The nodes' link capacities are chosen based on typical residential Internet connection types. We use the three capacity classes 25 MBit/s VDSL, 6 MBit/s ADSL, and 1 MBit/s ADSL (Table 9.4), equally distributed with a share of $\frac{1}{3}$ for each. The separation into node classes allows for an individual performance analysis for each of the classes. The network delays are based on the simulator's geographical delay model [86] with a global node distribution.

Both plain pSense and InterestCast are configured to use UDP as their transport protocol for messaging. This is a reasonable choice for an online gaming scenario and introduces the lowest possible packet header overhead. On the network layer, we assume IPv4. Hence, as small messages are comparably cheap with those protocols, we use optimistic assumptions for the direct dissemination of plain pSense. On the other hand, the dissemination algorithms have to cope with message loss, especially when links are saturated.



(a) All nodes

(b) Weak nodes only

Figure 9.9.: Relative bandwidth demand for a variation of node densities

The baseline application workload model is based on a gaming scenario, where each player regularly disseminates a state update. If not stated otherwise, updates have a payload of 48 bytes and are sent every 200 ms. This corresponds to a net data rate of 240 bytes/s per stream.

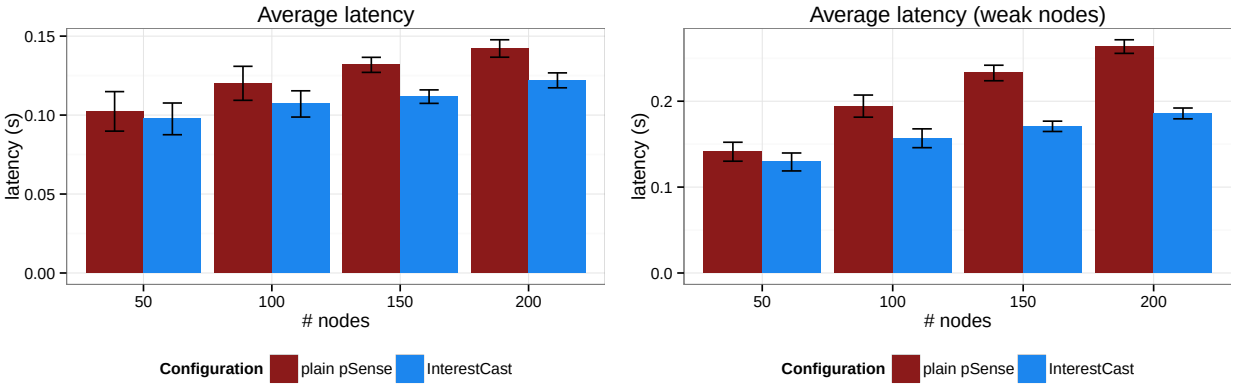
We stick to the same metrics as for the graph-based model wherever possible. Due to the different model, however, there are a few variations in the meaning of some metrics. Unlike in the graph-based model, the network simulator limits the maximum bandwidth that a node can transmit over its network link. To remain comparable with the graph-based model, the bandwidth demand metric counts all UDP datagrams that are supposed to be sent, even if they are immediately dropped at the local link due to congestion. Therefore, bandwidth demand can still grow beyond a value of 1. Yet, depending on the protocol, message loss can result in a reduced output rate, so that it is not strictly comparable with its counterpart in the graph-based evaluation. Further, latencies now include queuing delays in addition to the network propagation. With actual state information being transmitted, we can now measure staleness.

For the sake of clarity, we only show plots of the most important evaluation results in this section. Appendix C provides a collection of additional results.

9.4.1 Node Density

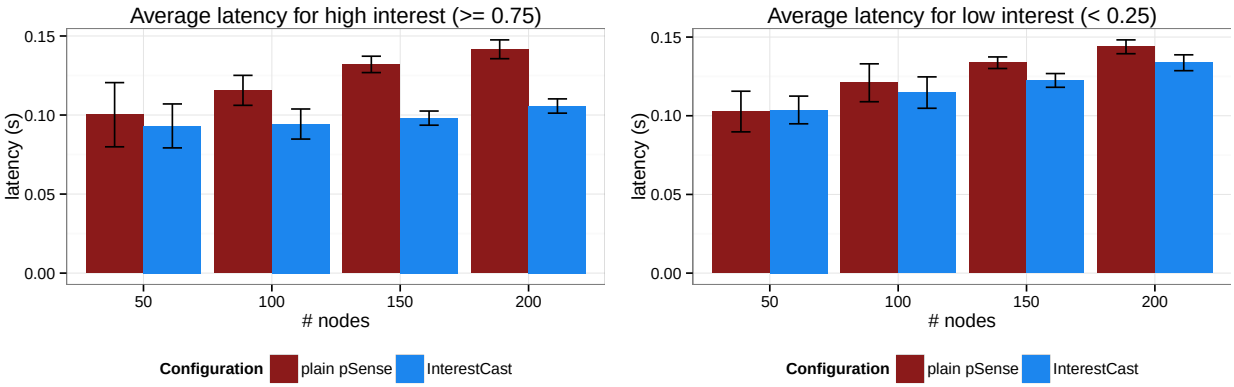
We first repeat the node density scenario from the graph-based model as described in Section 9.3.4. The node density is therefore varied between experiments, of which each uses static interest and network settings. We compare InterestCast together with pSense (‘InterestCast’) against plain pSense with its purely direct dissemination (‘plain pSense’).

Each experiment has a run time of ten minutes. In the first nine minutes, the nodes join and stabilize. At the same time, InterestCast’s optimization algorithm is already active. The last one minute is used for measurements of the converged state. The long stabilization phase is chosen pessimistically to work for all experiment sizes. In most situations, the system settles much quicker. The distribution of participants in the virtual world and the resulting interest set sizes are on average the same as for the graph-based scenario, see Table 9.2.



(a) All nodes **(b)** Weak nodes only

Figure 9.10.: Average latency for a variation of node densities



(a) High interest **(b)** Low interest

Figure 9.11.: Average latency by density of high and low interest neighbors

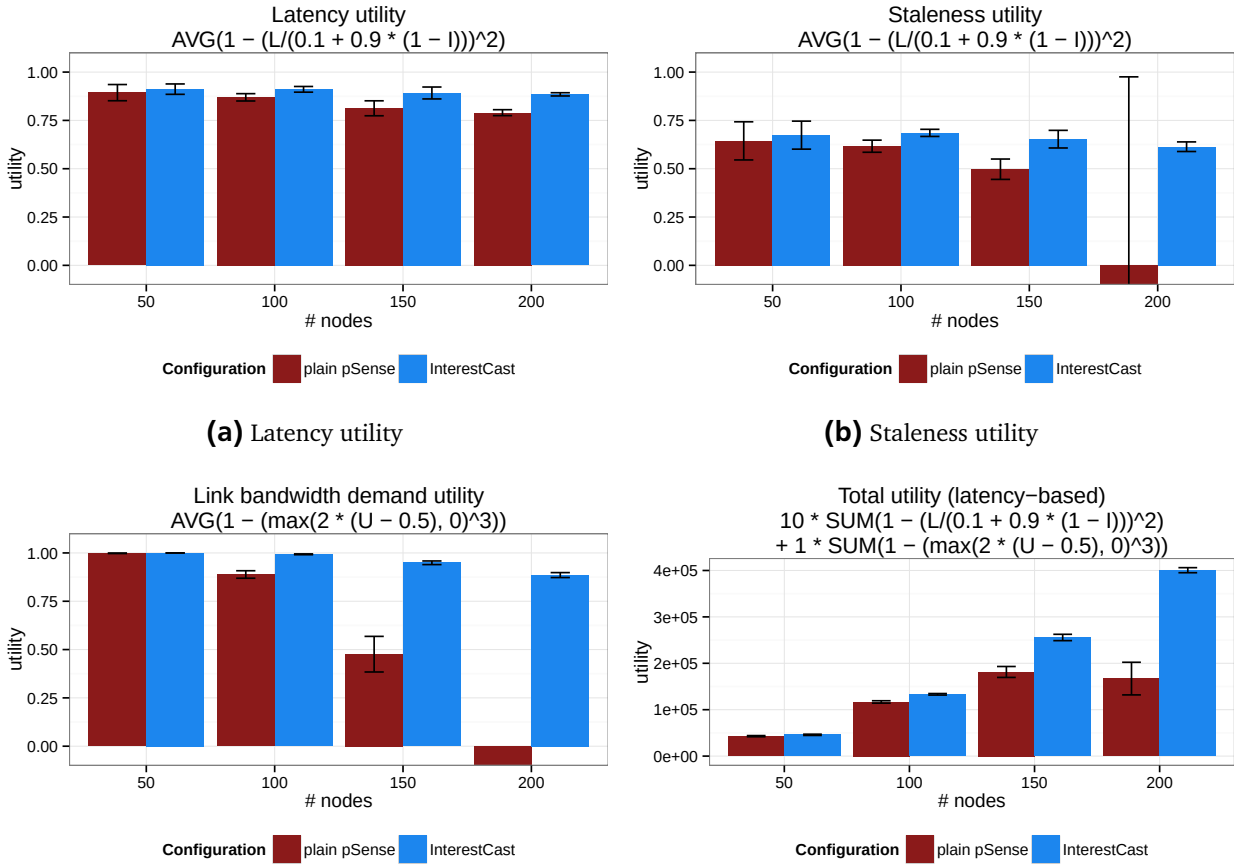
Bandwidth Demand

Figure 9.9 shows the bandwidth demand depending on the number of nodes. Like in the graph-based simulation, Interest significantly reduces the needed bandwidth. Looking only at the weak nodes (Figure 9.9b), the reduction is even more distinct. InterestCast almost allows doubling the number of nodes with the same average bandwidth demand as plain pSense.

Latency

The latency results (Figure 9.10) differ from the results of the graph-based model in that InterestCast beats plain pSense, allowing about a factor of two in node density for the same latencies. Although InterestCast uses longer paths, the reduced bandwidth demand, especially for the weak nodes, leads to a reduction in queuing delays, with the consequence of overall lower event delivery latencies. Queuing delays are not considered in the graph-based model. Again, the effect is most significant for weak nodes (Figure 9.10b).

Figure 9.11 differentiates event delivery latencies for high ($\iota \geq 0.75$) and low ($\iota < 0.25$) interest. Here, we can see that high-interest event delivery latencies barely suffer from an increased load



(a) Latency utility

(b) Staleness utility

(c) Bandwidth demand utility

(d) Total utility

Figure 9.12.: Utilities by density

due to higher densities. For high-interest events, InterestCast can handle a factor of almost four in node density without a significant effect on delivery latency. On the other hand, low-interest events are less prioritized and delivered via longer paths, so that they are affected by higher loads. Yet, due to the overall lower queuing delays, low-interest events have on average slightly lower latencies than with plain pSense, which supports no differentiation (e.g., prioritization) based on interest level.

Utility

The latency and bandwidth plots above show averages, hiding their distributions and hence the fairness among the nodes. The nonlinear utility functions take this into account and therefore serve as a better indicator for such, while at the same time reducing the overall results to a small and clear number of values.

The sums of utility according to the utility functions used are shown in Figure 9.12. Figures 9.12a and 9.12b show the *average* measured utilities for latency,

$$\text{i.e.,} \quad \frac{1}{|E|} \sum_{(v,u)\in E} \mathcal{U}_{\text{lat}}(\mathcal{L}_{u\to v}, I(v,u)),$$

and staleness,

$$\text{i.e.,} \quad \frac{1}{|E|} \sum_{(v,u)\in E} \mathcal{U}_{\text{lat}}(\mathcal{S}_{u\to v}, I(v,u)),$$

respectively. Figure 9.12c shows the average bandwidth utility

$$\frac{1}{|V|} \sum_{v\in V} \mathcal{U}_{\text{bw}}(b_v)$$

and Figure 9.12d shows the *total* weighted utility

$$w_{\text{lat}} \cdot \sum_{(v,u)\in E} \mathcal{U}_{\text{lat}}(\mathcal{L}_{u\to v}, I(v,u)) \; + \; w_{\text{bw}} \cdot \sum_{v\in V} \mathcal{U}_{\text{bw}}(b_v).$$

Due to the definition of the utility functions, the average utilities have a maximum of one, while the total utility is only bounded by the number of nodes and interest graph edges. Hence, the averages allow for a comparison between different numbers of nodes, while the total utility sum does not. There is no lower bound; in particular, utility sums can fall below zero. Such cases, however, are cut off in the plots because they indicate really bad results.

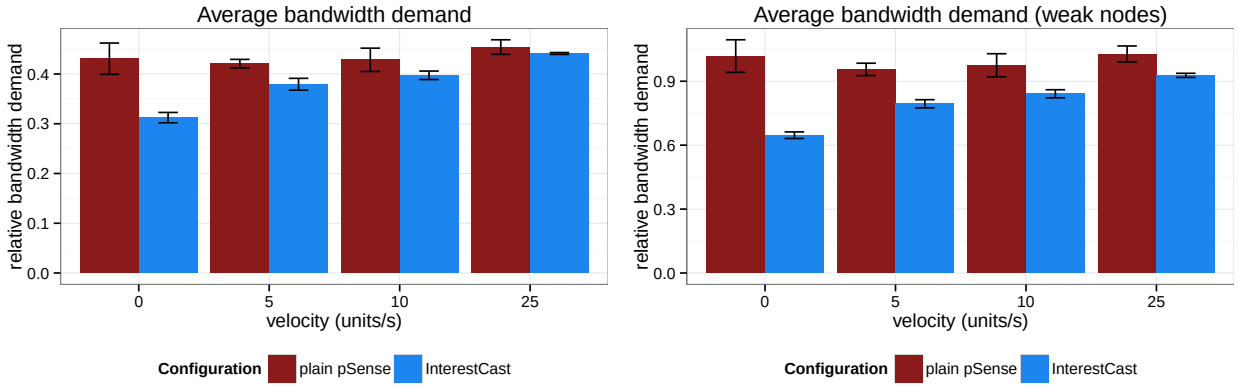
Starting at 150 nodes, plain pSense's bandwidth demands grow beyond the capacities, resulting in a quick degradation of the respective utility. The effect of the saturation and the resulting message loss becomes visible in the degrading staleness utility. InterestCast's utilities, on the other hand, remain much more stable.

9.4.2 Interest Dynamics

After analyzing InterestCast's behavior depending on the node density, we now focus on interest dynamics. Interest dynamics refer to both the change of interest levels as well as the consequent change of interest sets. Those dynamics have an impact on mainly two factors with respect to performance. First, the changes of neighbors and their interests require a continuous optimization, which never converges due to the continuous change. Secondly, frequent neighbor set changes lead to an increased rate of neighbor information updates and thus an increased management traffic overhead. The latter is further analyzed in Section 9.4.3.

For this experiment, we use the virtual reality scenario from above, with a few modifications. The interest dynamics are generated using a random waypoint mobility model. Each node selects a random point destination in the virtual world and moves towards it on a straight line with constant velocity. The destination points are uniformly distributed in the virtual world. This mobility pattern makes very pessimistic assumptions for InterestCast. Since all participants move totally independently, there are no groups that remain constant over a period of time, which would be beneficial for the optimization. Instead, neighbors most of the time just pass by each other, frequently entering and leaving each others' vision range.

The virtual world size is slightly increased to 1200×1200 units to compensate the fact that the eventual distribution of participants moving on the world is not uniform in this case. A higher concentration towards the center of the virtual world leads to a higher average number of interest set



(a) All nodes

(b) Weak nodes only

Figure 9.13.: Relative bandwidth demand depending on virtual world velocity

neighbors. For a better comparability, the interest set sizes are therefore reduced to approximately the same size as above by increasing the world size.

Keeping the number of nodes constant at 150, we scale the participants' velocities in the steps of 0, 5, 10, and 25 units per second. With a vision range of 150 units, a velocity of 25 units per second is already very high: Crossing each others' vision range takes at most 20 seconds.

As before, five experiments of ten minutes length each are run for each configuration. The measurements are averaged over the last minute of each experiment run. Error bars show the standard deviation between the experiment runs.

Bandwidth Demand

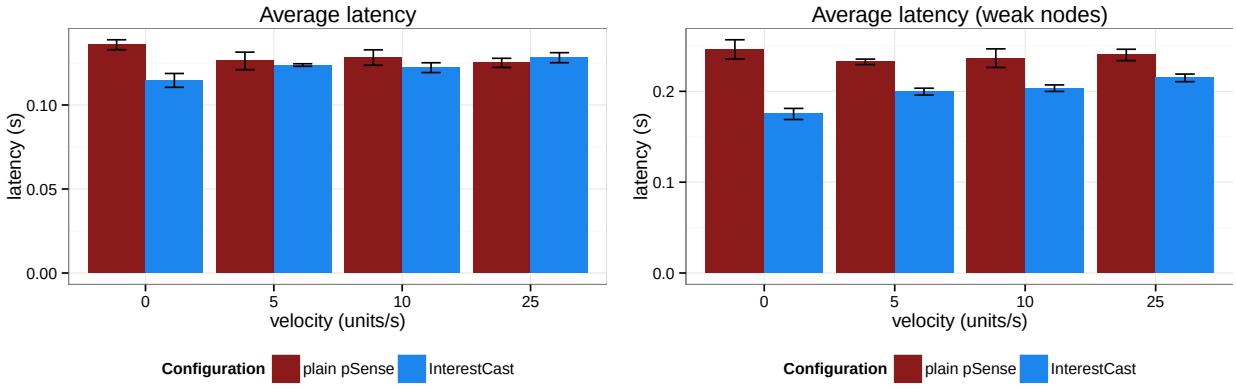
Figure 9.13 shows the bandwidth demand depending on the velocity. With higher interest dynamics, the distance to plain pSense shrinks. This is due to the management overhead of InterestCast, including first and foremost the neighbor information exchange and the routing update operations. Both need to be performed more frequently with higher velocities and therefore become more expensive in terms of bandwidth overhead. For weak nodes, however, there remains a significant reduction in bandwidth demand, improving load fairness.

Latency

The decrease of InterestCast's efficiency with high dynamics is also visible in the latency measurements (Figures 9.14 and 9.15). Yet, for weak nodes as well as for high interest neighbors, the latencies remain significantly better than without InterestCast.

Utility

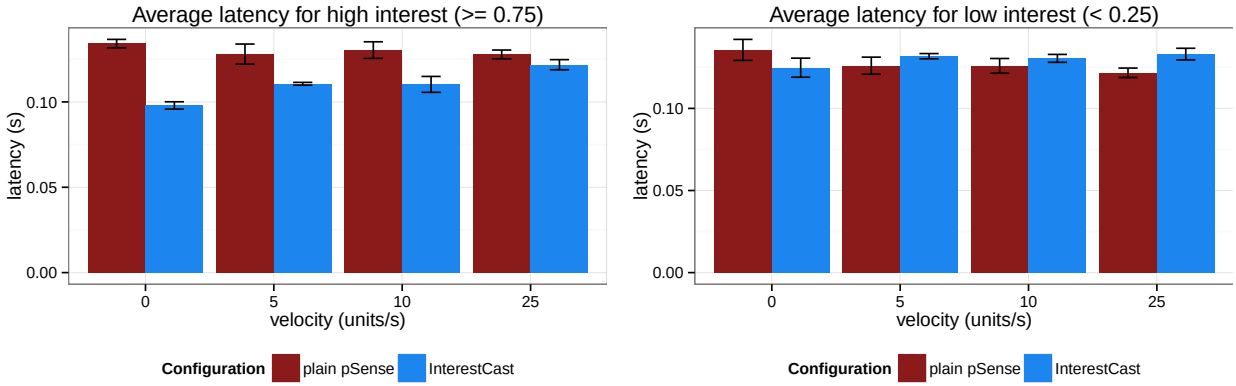
The utility sums based on the selected utility functions are shown in Figure 9.16. They as well show the consistently better performance of InterestCast, except for very high dynamics.



(a) All nodes

(b) Weak nodes only

Figure 9.14.: Average latency depending on virtual world velocity



(a) High interest

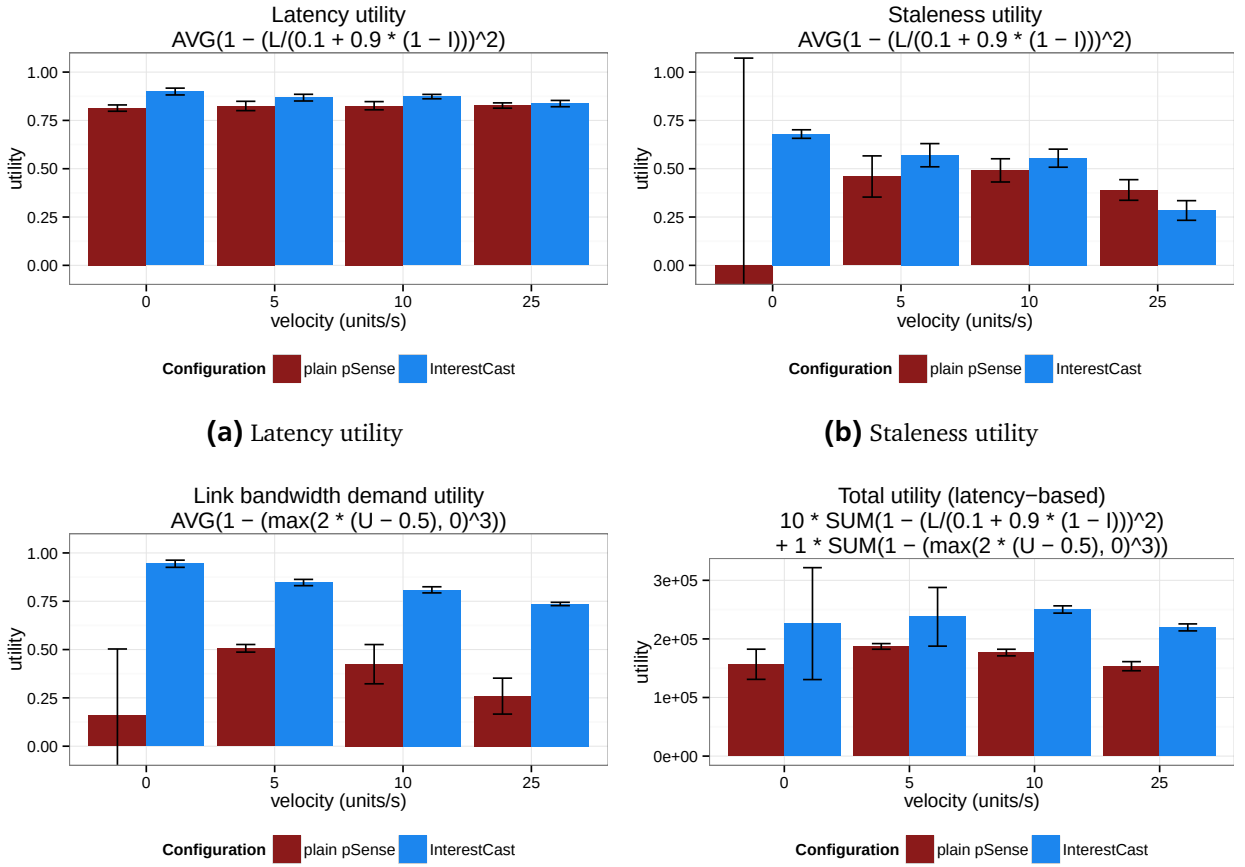
(b) Low interest

Figure 9.15.: Average latency by velocity of high and low interest neighbors

Missing Neighbors

As a final aspect of this section, we look at missing neighbors. Missing neighbors are those participants that are of interest, i.e., are within the vision range, but are not visible to the application. Although the discovery of neighbors is the job of the interest management, which is not in the focus here, the reliability of the event dissemination plays an important role. In addition, missing neighbors are not considered in the latency measurements. They further reduce the effective set of participants to which the events are delivered. This reduces the bandwidth demand and must therefore be considered.

Figure 9.17 shows the ratio of missing neighbors. It clearly shows that with InterestCast, the missing neighbors ratio is drastically lower. The neighbor discovery and introduction process works much more reliably with InterestCast, being a clear indication of the better capacity management. Only with very high dynamics, a certain ratio of missing neighbors is inevitable due to the discovery and introduction process, which delays the connection to new neighbors entering the vision range.



(a) Latency utility

(b) Staleness utility

(c) Bandwidth demand utility

(d) Total utility

Figure 9.16.: Utilities by velocity.

9.4.3 Traffic Overhead

A critical factor affecting the overall performance of InterestCast is the communication overhead of management data. This management data includes route operation messages as well as the neighbor information exchange messages for the mutual state monitoring. While the traffic generated by the former is marginal, the latter can have a significant impact. As analyzed in Section 6.3, large neighbor sets and high ratios of common neighbors due to a high clustering lead to a growth in neighbor state that needs to be exchanged. The exchange frequency demand further grows with the interest dynamics. We therefore measure the traffic broken down by message type for the relevant messages for varying virtual world velocities.

Figure 9.18 shows the traffic breakdown. The first four message types are InterestCast management messages, while the last two are application messages from InterstCast's point of view.

- BandwidthMessages contain information on the nodes' available and used bandwidth.
- Known neighbors are exchanged as bloom filters in NeighborFilterMessages.
- LatencyMessages contain the latencies to common neighbors.

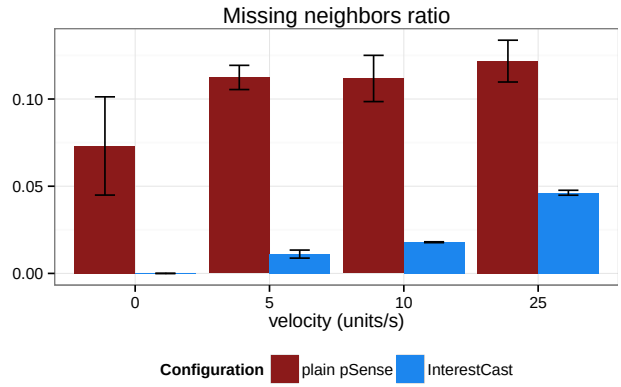


Figure 9.17.: Missing neighbors ratio ($1 - \frac{\text{neighbors in vision range}}{\text{known neighors}}$) of plain pSense and InterestCast depending on the velocity

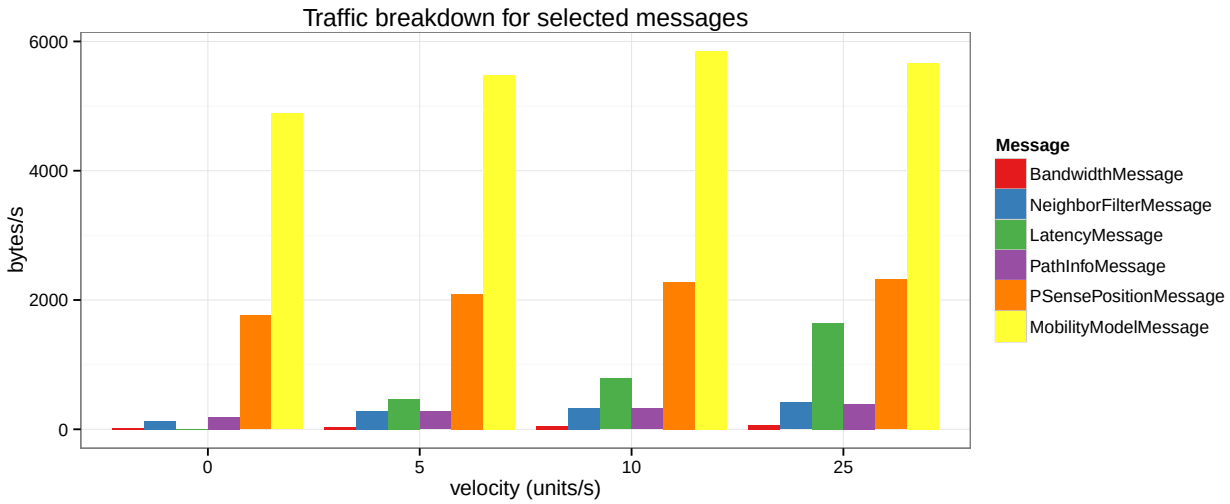
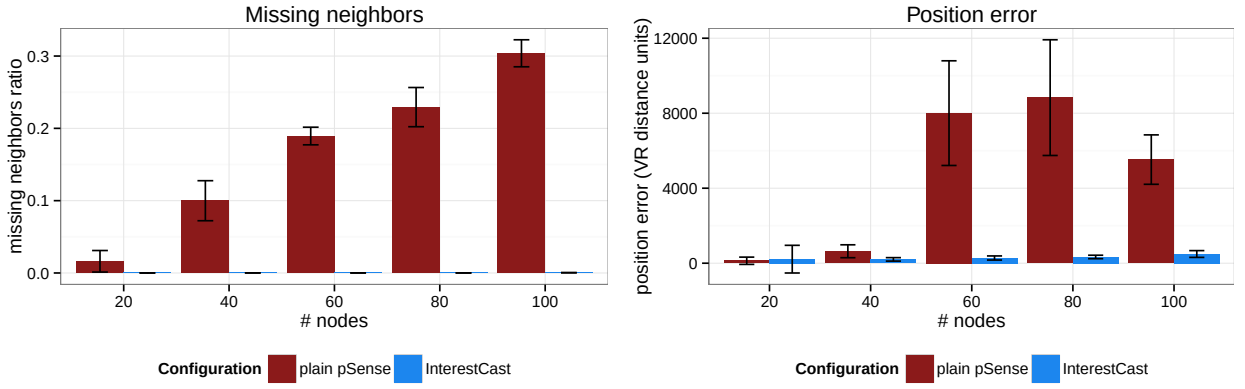


Figure 9.18.: Average per-node message traffic for a selection of important message types

- PathInfoMessages are regularly transmitted backwards along each path to communicate the number of hops and total latency.
- PSensePositionMessages are low-frequency position updates of the pSense interest management.
- The actual application payload, finally, is in this case transmitted as MobilityModelMessages.

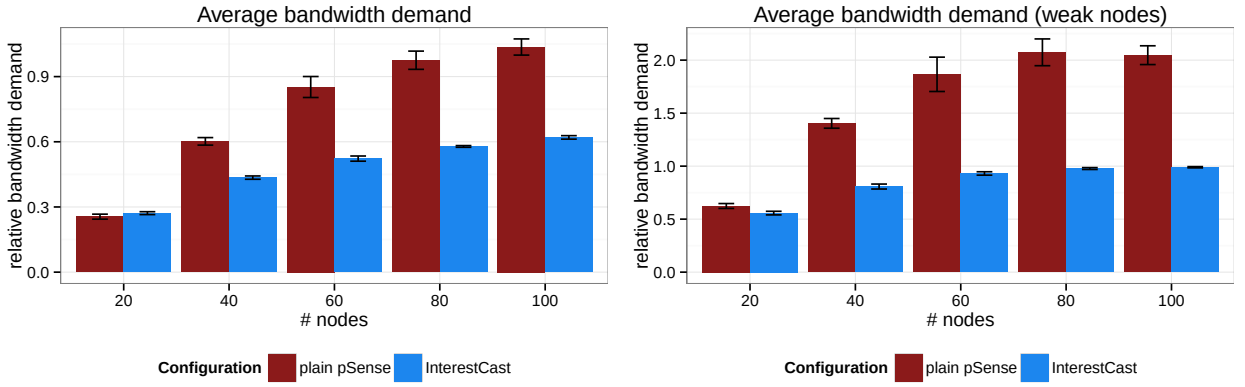
The average number of neighbors varies slightly depending on the velocity, which explains the variations in application and interest management traffic. One can observe that the traffic volume of BandwidthMessage, NeighborFilterMessage, and PathInfoMessage barely grow with the velocity. LatencyMessages, on the other hand, take a significant proportion of the total traffic in cases of high velocities. This can be explained with the quadratic growth of latency information depending on the number of common neighbors (cf. Section 6.3). Hence, for very high dynamics, the management overhead can outgrow the bandwidth savings from the optimization. For medium velocities, however, the overhead remains moderate.



(a) Missing neighbors ratio

(b) Position error

Figure 9.19.: Error metrics with Planet PI4 bot workload depending on virtual world density



(a) All nodes

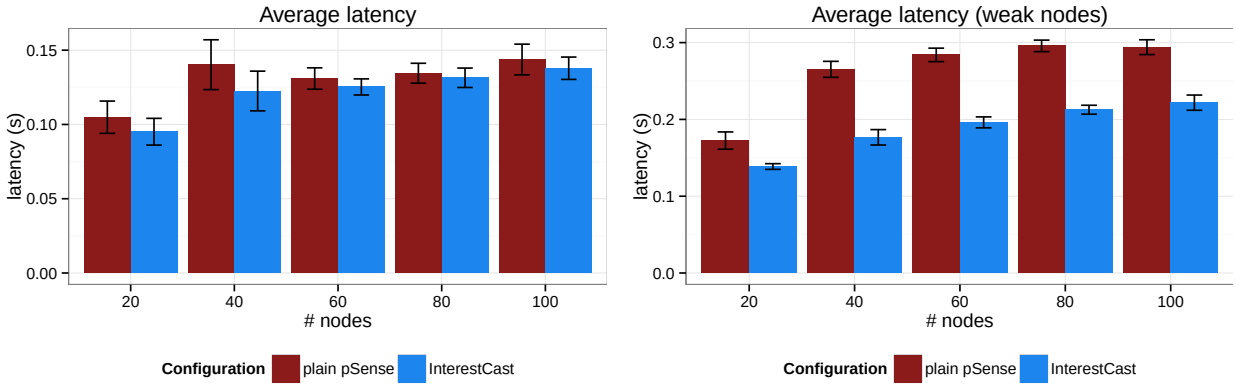
(b) Weak nodes only

Figure 9.20.: Relative bandwidth demand with Planet PI4 bot workload depending on virtual world density

9.4.4 Planet PI4 Bot Workload

So far, we have used synthetic workloads based on a mobility model because they allow fine-grained control over the parameters and low-level metrics such as delivery latency and staleness. To show the feasibility in a real gaming scenario, we use the Planet PI4 gameplay (cf. Chapter 8) played by bots [47]. The bots provide a reproducible and scalable workload on top of the game [103]. According to the higher-level workload, we also consider higher-level metrics. Most important is the *position error* [69, 99], which measures the position difference of a player's *real* position and the positions of this player as they are *perceived* by its neighbors at a given point in time. The real position is considered to be the position that the respective player perceives of itself. Hence, the position error is the higher-level metric related to staleness and virtual world velocity.

In the experiments, we scale the number of players and thereby the virtual world density. Due to the flocking behavior of the bots, the resulting density for a given number of players is higher than



(a) All nodes

(b) Weak nodes only

Figure 9.21.: Average latency with Planet PI4 bot workload depending on virtual world density

before. In addition, the events being generated are not only regular position updates, but also shot and hit events. This results in a further workload increase.

Figure 9.19 shows the ratio of missing neighbors as well as the average position errors. As shown above with a mobility model workload, the number of missing neighbors (Figure 9.19a) grows rapidly with plain pSense due to the network overload. InterestCast is much more stable here. The picture is similar for the position error (Figure 9.19b). Starting at 60 players, it grows dramatically with plain pSense, only mitigated by the high numbers of missing neighbors. InterestCast, on the other hand, only shows a moderate increase, thus handling the situation significantly better. Like for the mobility model workloads, InterestCast's bandwidth demand is lower (Figure 9.20) and the latency of the events particularly from week nodes is significantly improved (Figure 9.21).

9.5 Discussion

For the evaluation of InterestCast, we chose a two-stage model. The first uses highly abstracted graph-based models for interest and network properties. With those, our optimization problem is solved in two variants: using a standard linear programming solver to obtain global optima and with InterstCast's incremental local optimization algorithm. In the second stage, the full protocol implementation of InterestCast is tested in a detailed network simulation and compared against plain pSense.

In the first stage we have shown that InterestCast's incremental local optimization, although not providing global optima, provides reasonably competitive solutions. In this context it needs to be stressed that the global solutions can only serve as a benchmark. Implementing a global solver is expensive in runtime, as we have shown, and would further require that all participants transmit their state (bandwidth, latencies, etc.) to one single place where the optimization is done. The advantages of the local algorithm therefore clearly outweigh the slightly suboptimal results.

The full protocol implementation tested in the second stage includes all the necessary monitoring and route management infrastructure. Hence, the first thing that the evaluation shows is that the concept can be practically implemented in a fully distributed system. More important, however, is

the comparison with plain pSense, which uses direct dissemination only but also gets along without the additional monitoring and management overhead. In low demand situations, InterestCast shows to be competitive, mostly relying on direct communication as well. In high load situations, where especially weak nodes are overloaded with direct dissemination, InterestCast significantly improves the results. The higher propagation delays due to the indirection are by far compensated by the reduced queuing and message loss rates. Furthermore, InterestCast connects neighbors more reliably by mitigating load peaks and thereby achieves significantly lower missing neighbor rates.

With higher interest dynamics, it becomes harder to complete against the purely direct dissemination. Interest dynamics lead to an increased overhead traffic due to InterestCast's neighbor information exchange, which is not needed for plain pSense. In addition, InterestCast's incremental optimization does never converge due to the constant change and therefore does not reach its best state. Both factors have a negative impact in InterestCast's performance, so that at some point the direct dissemination becomes the best choice. An option to improve the situation if very high dynamics need to be supported could be to switch to a plain direct dissemination in such cases. One final point that potentially adds to the reduced performance with dynamics is the greedy optimization based on the redirect and shortcut operations. The optimization due to changed interest may require to leap local optima much more often than the initial optimization starting from a direct dissemination. Hence, the optimization might get stuck in the local optima more often.

It has to be noted, however, that the mobility model used to generate interest dynamics is a very pessimistic choice. In many cases of virtual world scenarios, as represented in this evaluation by the bot workloads, groups of participants move together such that intra-group dynamics are significantly lower. In those cases, InterestCast is clearly superior to plain pSense, especially with respect to the performance of weak nodes.

10 Conclusion and Outlook

In this thesis, we have developed InterestCast, a fully distributed event dissemination system, which adapts to both the application needs and network conditions. To achieve this, we first elaborated on potential target applications, focusing on online gaming. Based on those, we identified the main requirements and challenges. After discussing existing solutions and their shortcomings, we derived the basic design decisions: the decomposition of interest management and event dissemination and their interfacing, the basic adaptation concepts, InterestCast's routing, and its optimization. Afterwards, we formalized our system model and formulated the optimization goal. We detailed InterestCast's core algorithm, the local incremental optimization. The full prototype protocol implementation as well as the framework Planet PI4, in which the implementation and evaluation took place, were described, before the evaluation with a two-staged model was presented.

The main goal of this work was a many-to-many event dissemination with a particular focus on low dissemination latencies. Our proposed solution, InterestCast, is able to adapt with respect to both application needs and network conditions. Further, it gains from heterogeneous environments and needs no dedicated infrastructure, i.e., works in a decentralized and self-organizing manner. InterestCast starts with a direct dissemination scheme and continuously runs a local optimization algorithm that modifies event routes to use neighboring nodes as forwarders where suitable. This way, it makes use of data aggregation and reduces the number of small packets with high packet header overhead. Application needs are expressed as utility functions, which in the concrete case consider bandwidth demand as well as latency depending on the interest level.

InterestCast provides a distributed event dissemination, while it relies on existing interest management solutions, such as pSense, which works fully decentralized as well. To make this possible, we needed a clear separation of functionality between interest management and event dissemination. InterestCast's interface to the interest management component uses interest levels as an application-independent abstraction of interest gradations. This is an extension to the conventional "interest or no interest" subscription model.

The optimization of the system state based on utility functions is a good fit for best-effort solutions. Where no hard guarantees can be given, the utility-based optimization is able to make trade-offs based on the particular application needs. Using additive utilities, it is possible to predict effects on overall utility based on local partial knowledge. Further, non-linear utility functions allow considering fairness.

The comparison with global optima based on a graph-based network model has shown that the local optimization provides good results compared to the global optima. Computing global optima at runtime, on the other hand, is infeasible as it would require all information about the current network and load situation at a single point. It has further shown to be too expensive to compute for any relevant network size, despite the use of simplistic network models.

In detailed network simulations, the InterestCast prototype, which is suitable for real-world deployments, has proven its superiority over a plain direct dissemination. Especially in situations of high load, it significantly reduces the bandwidth demand of overloaded nodes and thereby also decreases dissemination latencies. It allows doubling the node density and still provides the same average latency than plain pSense. Hence, InterestCast's performance is improved by the likely heterogeneity of the participants' devices and connections. Further, high-interest events have the lowest latency penalty, demonstrating how the interest level can be used to prioritize events. For high-interest events, the latency remains almost constant even with a node density factor of four.

InterestCast was evaluated with scenarios adopted from multiplayer online games. Those have high demands particularly with respect to latency, which are well studied. This makes them well suited for our evaluation purposes. The core results can be generalized to the other discussed application scenarios as well. With respect to the workloads, the gaming scenarios have similar magnitudes as expected in coordination tasks of robotics or vehicular networks. Ad-hoc networking scenarios require an adapted system model and work in different latency ranges than global-scale networks, but are otherwise comparable.

We have developed a fundamental concept for a decentralized utility-based optimization of a many-to-many event dissemination middleware, which can be adapted to related problems using factors beyond latency and bandwidth demand. With a system model that can be extended in a modular fashion, it allows for further advances in the abstraction of the application utility functions from the underlying system.

10.1 Future Work

With InterestCast, we have developed a fundamental concept for a decentralized utility-based optimization of a many-to-many event dissemination middleware. Yet, there remains room for future extensions and enhancement in several aspects. In this section, we discuss those, starting with incremental additions and moving towards more sophisticated additions.

Computational Efficiency

The current optimization algorithm was developed with the goal of a low communication and state demand, but does not take into account the computational effort. Although this is not a significant factor in many cases, the brute force enumeration of all possible transitions in each optimization iteration can become a limiting factor with a large number of neighbors. Hence, a more efficient algorithm that works without a full enumeration of options can improve the situation in such case. One approach could be to use heuristics and to decide based on partial evaluation of the utility impact of possible transitions. Other options are to parallelize the evaluation of possible transitions or to re-compute only the options whose expected utility impact has changed.

Dynamic Optimization Rate

For both reducing necessary computation and increasing optimization speed, a dynamic rate adaptation for the incremental optimization can be introduced. With such, the optimization steps would be invoked more frequently under high dynamics and less frequently under low dynamics.

Since each node runs the optimization individually, there is no need for a coordination of optimization rates. The rate adaptation could be done based on the interest and neighbor set dynamics or the achieved utility improvements of the previous optimization steps.

Specializations for Update Events

The delivery of update events, which contain updates of continuously changing state such as positions, can be improved by adding specific handling of those. In the current solution, for the sake of generality, we assume that all events pushed by the application should be delivered instantly. Update events on the other hand could be pulled from the application with fresh state whenever the event dissemination decides to send one. This could be used to fill up existing packets, but also allows the event dissemination mechanism to adapt the update rate. This allows further taking into account the staleness utility in the optimization.

Utility-Based Scheduling

A more generalized version of the above would be to consider the utility functions in the scheduling algorithm for all events. This way, prioritization of events in high load situations could adapt better to the application needs. The scheduler can use the accumulated latency that is already included in every message to heuristically select the forwarding priority based on the remaining timeframe. The interaction between the utility-based routing and scheduling, however, has to be considered carefully to avoid unintended feedback loops.

Generalization of the Local Optimization Approach

InterestCast implements the local incremental optimization approach for a specific purpose. The basic concept, however, is applicable in a much broader sense. A first extension would be the addition of further route operations, such as directly switching a forwarder node instead of applying a shortcut and redirect operation. This can reduce the potential of local optima that hinder the optimization. On the other hand, however, such operation requires neighbor information to be exchanged one hop further, which introduces additional costs. Yet, the neighbor information exchange algorithm can be generalized to support multiple hops where needed by the used operations.

Additional Utility Factors

InterestCast works with the utility factors latency, interest, and bandwidth demand. The basic optimization concept, however, makes it easy to add additional factors. For instance, a reliability factor can be introduced, if the system allows for a redundant transmission of data or spare paths for failover.

Consideration of Special Network Properties

InterestCast's current model only considers one network interface per node, which has been the case for most end user Internet devices. Recently, however, an increasing number of devices have multiple network interfaces, such as WiFi and cellular broadband. While they were initially only used alternatively, using multiple interfaces at once opens a variety of new use cases. One

example is mobile connection sharing [125]. Considering multiple interfaces and their individual properties and costs separately, InterestCast could provide similar features, such as using certain nodes as forwarding gateways between the cellular network and an ad-hoc WiFi network.

Inclusion of Active Network Elements

The local optimization provides potential for the inclusion of locally available active network elements, such as dynamically allocated virtual machines in network routers. Although they are still uncommon in real network deployments, active research in cloudlets [141] and network function virtualization (NFV) [81] may promote their future availability. With those, dedicated InterestCast nodes could be dynamically deployed in the network, allowing for a more efficient yet flexible event dissemination.

A Global Optimization Integer Programs

This section lists and briefly explains the complete integer programs described in Section 5.6.

A.1 ILP Problem

The following listing contains the Zimpl program of the global optimization problem simplified to an ILP program (cf. Section 5.6.1). Node IDs, link capacities, as well as X any Y coordinates for the delay model are stored in the columns of file *nodes.dat*. Interest levels between pairs of nodes are stored in *interest.dat*.

```
1  # nodes
2  set V := { read "nodes.dat" as "<1n>" comment "#" };
3
4  # node bandwidths in #flows
5  param bw[V] := read "nodes.dat" as "<1n> 2n" comment "#";
6
7  # nodes' geo corrdinates
8  param geo_x[V] := read "nodes.dat" as "<1n> 3n" comment "#";
9  param geo_y[V] := read "nodes.dat" as "<1n> 4n" comment "#";
10
11 # interest of 1st node in 2nd node
12 param interest[V*V] := read "interest.dat" as "<1n, 2n> 3n" comment"#";
13
14 # minimum per-link latency
15 param baseLatency := 0.01;
16 # latency per geo-distance unit
17 param geoLatencyFactor := 0.002;
18
19 # subscription of l at k, i.e., flow from k to l
20 set subscriptions := { <k,l> in V*V with (k != l) and (interest[l,k] > 0) };
21
22 # latency model
23 defnumb latency(a,b) := sqrt((geo_x[a] - geo_x[b])^2 + (geo_y[a] - geo_y[b])
       ^2) * geoLatencyFactor + baseLatency;
24
25 # variables
26
27 # y[i,j,k,l] = 1 iff events for flow from k to l are routed from i to j
28 var y[V*V*V*V] binary;
29 # helper variables to achieve maximum operation:
30 # x[i,j,k] = 1 iff events for flow from k are routed from i to j
```

```
var x[V*V*V] binary;
# z[i,j] = 1 iff events are routed from i to j
var z[V*V] binary;

# objective function

minimize cost:
  sum <i,j,k,l> in V*V*V*V: (y[i,j,k,l] * latency(i, j));

# constraints

# if at least one flow from k to some l is routed from i to j,
# there is a flow from k routed from i to j
subto x_y:
  forall <i,j,k,l> in V*V*V*V:
    x[i,j,k] >= y[i,j,k,l];
# if at least one flow from some k is routed from i to j,
# there is a flow routed from i to j
subto z_x:
  forall <i,j,k> in V*V*V:
    z[i,j] >= x[i,j,k];

# bandwidth limit
# counting all flows from individual sources (x) and connection overhead (z)
subto bwbound:
  forall <i> in V:
    (sum <j,k> in V*V: x[i,j,k] + sum <j> in V: z[i,j]) <= 1.0 * bw[i];

# in-degree == out-degree, except for source and destination
subto path:
  forall <k,l> in subscriptions:
    forall <i> in V without { <k>, <l> }:
      (sum <j> in V: y[j,i,k,l]) == (sum <j> in V: y[i,j,k,l]);

# nodes don't route to themselves
subto no_self_loop:
  forall <i> in V:
    z[i,i] == 0;

# source has exactly one outgoing flow
subto source_out:
  forall <k,l> in subscriptions:
    sum <j> in V: y[k,j,k,l] == 1;
subto source_no_in:
  forall <k,l> in subscriptions:
    sum <j> in V: y[j,k,k,l] == 0;
```

```
77
78 # destination has exactly one incoming flow
79 subto destination_in:
80   forall <k,l> in subscriptions:
81     sum <i> in V: y[i,l,k,l] == 1;
82 subto destination_no_out:
83   forall <k,l> in subscriptions:
84     sum <i> in V: y[l,i,k,l] == 0;
```

A.2 MINLP Problem

The following listing extends the ILP problem to a MINLP problem, as discussed in Section 5.6.2. Due to limitations in the Zimpl language, the cost functions (C_{lat}, C_{bw}) cannot be directly represented in the program code in all cases. Instead, it might be necessary to introduce additional helper variables, e.g., for maximum and minimum operations.

Here, we use the exemplary cost functions (cf. Section 6.5)

$$C_{\text{lat}}(\mathscr{L},\iota) = \left(\frac{\mathscr{L}}{0.1+0.9*(1-\iota)}\right)^2 \quad \text{and}$$
$$C_{\text{bw}}(b) = \max(4\cdot(b-0.5),0)^3 .$$

```
1  # nodes
2  set V := { read "nodes.dat" as "<1n>" comment "#" };
3
4  # node bandwidths in #flows
5  param bw[V] := read "nodes.dat" as "<1n> 2n" comment "#";
6  param bw_factor := 1.0;
7
8  # nodes' geo corrdinates
9  param geo_x[V] := read "nodes.dat" as "<1n> 3n" comment "#";
10 param geo_y[V] := read "nodes.dat" as "<1n> 4n" comment "#";
11
12 # interest of 1st node in 2nd node
13 param interest[V*V] := read "interest.dat" as "<1n, 2n> 3n" comment "#";
14 param interestThreshold := 0.1;
15
16 # minimum per-link (last mile) latency
17 param baseLatency := 0.01;
18 # latency per geo-distance unit
19 param geoLatencyFactor := 0.002;
20
21 param weightPathLatency := 1.0;
22 param weightBandwidth := 64.0;
23
```

```
24 # subscription of l at k, i.e., flow from k to l
25 set subscriptions := { <k,l> in V*V with (k != l) and (interest[l,k] >=
       interestThreshold) };
26
27 defnumb latency(a,b) := sqrt((geo_x[a] - geo_x[b])^2 + (geo_y[a] - geo_y[b])
       ^2) * geoLatencyFactor + baseLatency;
28
29 # variables
30
31 # y[i,j,k,l] = 1 iff events for flow from k to l are routed from i to j
32 var y[V*V*V*V] binary;
33
34 # helper variables to achieve maximum operation:
35 # x[i,j,k] = 1 iff events for flow from k are routed from i to j
36 var x[V*V*V] binary;
37 # z[i,j] = 1 iff events are routed from i to j
38 var z[V*V] binary;
39
40 # helper variables for non-linear latency cost
41 # subscription latency cost (based on non-linear utility function)
42 var c_lat[subscriptions];
43 # node bandwidth cost (based on non-linear utility function)
44 var c_bw[V];
45 # node bandwidth maximum helper
46 var d_bw[V];
47 # relative node bandwidth demand
48 var u_bw[V];
49
50 # objective function
51
52 minimize cost:
53   (sum <k,l> in subscriptions: c_lat[k,l] * weightPathLatency)
54   + (sum <i> in V: c_bw[i] * weightBandwidth);
55
56 # constraints
57
58 # c_lat: latency cost
59 # cost function: 1 / (0.1f + 0.9f * (1.0f - i))
60 subto c_lat:
61   forall <k,l> in subscriptions:
62     c_lat[k,l] == (sum <i,j> in V*V: y[i,j,k,l] * latency(i, j)) / (0.1 + 0.9
         * (1.0 - interest[k,l]));
63
64 # c_bw: bandwidth cost
65 # cost function: max(4 * (b - 0.5), 0)^3
66 subto c_bw:
```

```
67    forall <i> in V:
68      c_bw[i] == d_bw[i]^3;
69  # max(4 * (b - 0.5), 0)
70  subto d_bw:
71    forall <i> in V:
72      d_bw[i] >= 4 * (u_bw[i] - 0.5);
73  subto d_bw_gz:
74    forall <i> in V:
75      d_bw[i] >= 0;
76  # relative bw demand
77  # counting all flows from individual sources (x) and connection overhead (z)
78  subto u_bw:
79    forall <i> in V:
80      u_bw[i] == (sum <j,k> in V*V: x[i,j,k] + sum <j> in V: z[i,j]) / bw[i];
81
82  # if at least one flow from k to some l is routed from i to j,
83  # there is a flow from k routed from i to j
84  subto x_y:
85    forall <i,j,k,l> in V*V*V*V:
86      x[i,j,k] >= y[i,j,k,l];
87  # if at least one flow from some k is routed from i to j,
88  # there is a flow routed from i to j
89  subto z_x:
90    forall <i,j,k> in V*V*V:
91      z[i,j] >= x[i,j,k];
92
93  # in-degree == out-degree, except for source and destination
94  subto path:
95    forall <k,l> in subscriptions:
96      forall <i> in V without { <k>, <l> }:
97        (sum <j> in V: y[j,i,k,l]) == (sum <j> in V: y[i,j,k,l]);
98
99  # nodes don't route to themselves
100 subto no_self_loop:
101   forall <i> in V:
102     z[i,i] == 0;
103
104 # source has exactly one outgoing flow
105 subto source_out:
106   forall <k,l> in subscriptions:
107     sum <j> in V: y[k,j,k,l] == 1;
108 subto source_no_in:
109   forall <k,l> in subscriptions:
110     sum <j> in V: y[j,k,k,l] == 0;
111
112 # destination has exactly one incoming flow
```

```
113 subto destination_in:
114   forall <k,l> in subscriptions:
115     sum <i> in V: y[i,l,k,l] == 1;
116 subto destination_no_out:
117   forall <k,l> in subscriptions:
118     sum <i> in V: y[l,i,k,l] == 0;
```

B InterestCast Routing Update Handling

This section details the operations performed for InterestCast's route update operations, as described in Section 7.3.1, using flowcharts. The flowcharts describe the basic operations upon reception of update messages via the procedures onRouteUpdate, onRouteNotify, and onRouteAck, respectively, in Figures 7.5 and 7.6.

onRouteUpdate, depicted in Figure B.1, handles HOP_ROUTE_UPDATE requests. In Figures 7.5 and 7.6, this procedure is invoked as the first operation on U and S, respectively. As parameters, it gets the source-destination tuple (S', D'), which identifies the route, as well as the new via node (via) and the route ID RID that will be used by the predecessor in the following. The procedure checks the feasibility of the operation and, if successful, forwards a HOP_ROUTE_NOTIFY to the new via node (via).

Subsequently, onRouteNotify (Figure B.2) is triggered on that via node. This procedure checks the applicability from the point of view of the via node and replies with a HOP_ROUTEACK(ACCEPT), if successful.

The HOP_ROUTEACK message is processed by onRouteAck (Figure B.3). HOP_ROUTEACK messages serve as responses to both HOP_ROUTE_UPDATE and HOP_ROUTE_NOTIFY. The handler therefore uses the pending operation registry to determine the operation associated with a particular HOP_ROUTEACK message. A special case is the timeout (indicated as $answer = \text{TIMEOUT}$), which is the absence of an acknowledgement within a certain time frame.

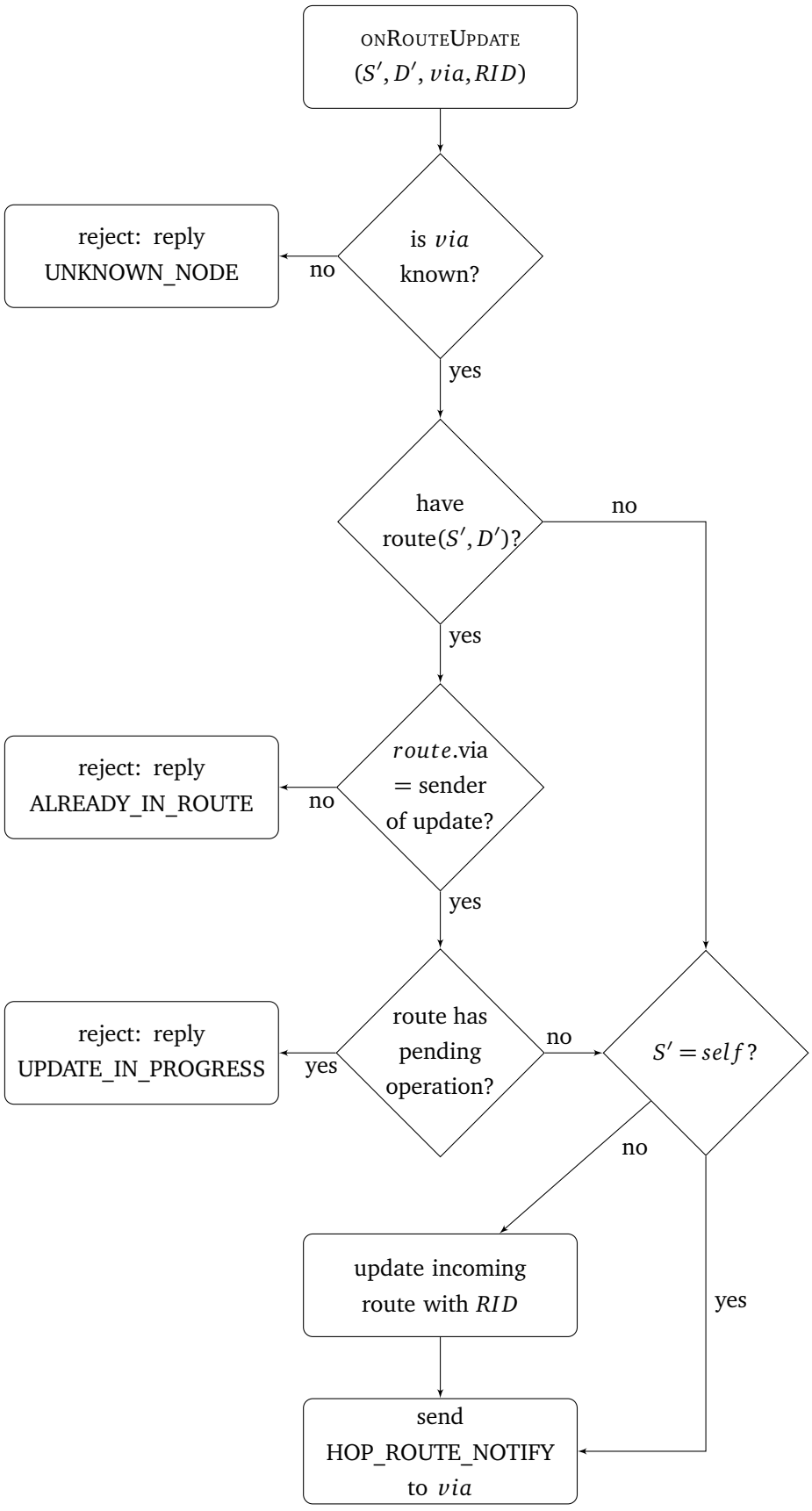


Figure B.1.: Flowchart for the handling of a route update message

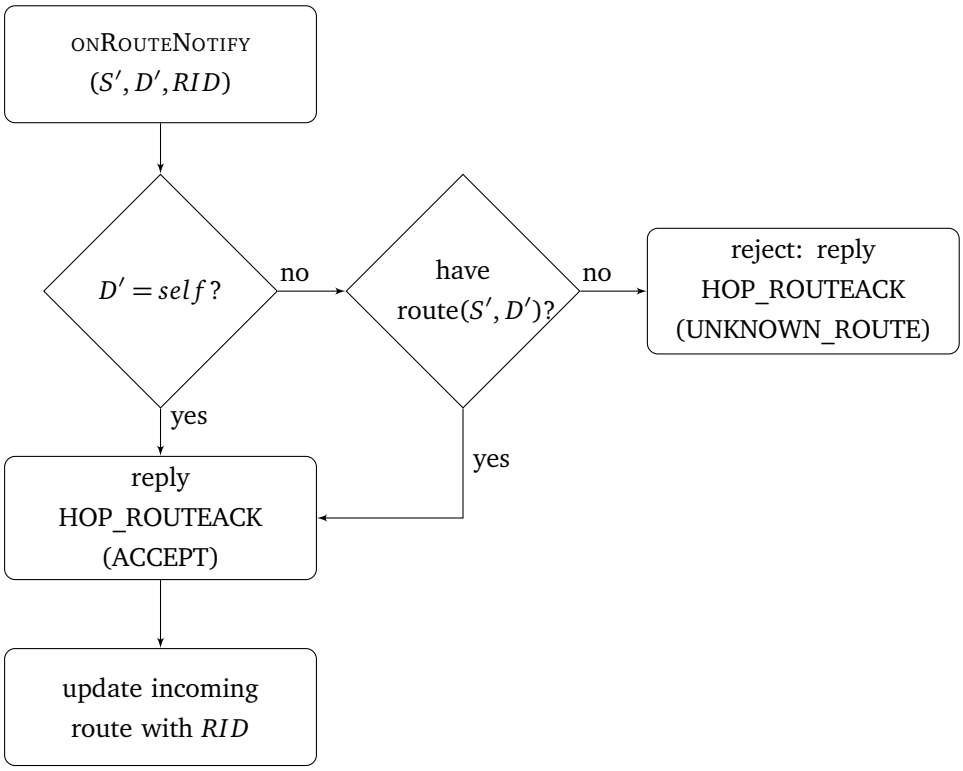


Figure B.2.: Flowchart for the handling of a route notify message

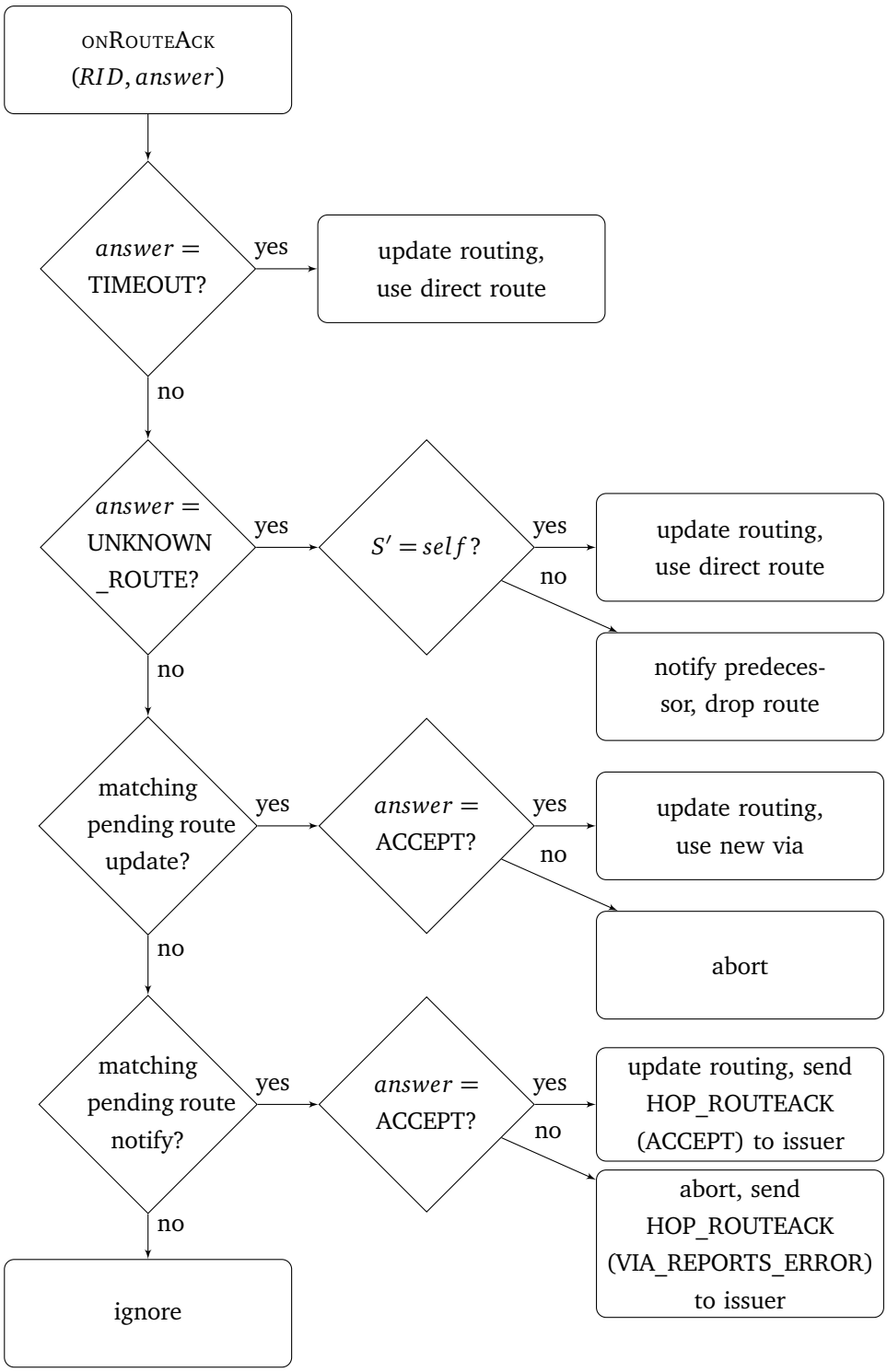


Figure B.3.: Flowchart for the handling of a route update acknowledgement message

C Prototype Evaluation Results

This appendix section provides a more complete compilation of evaluation results of the InterestCast prototype that were omitted for reasons of clarity in Chapter 9.

Figures C.1, C.2, C.3, and C.4 show the results of the virtual world density variation experiment, systematically separated by node capacity class and interest level. The properties of the capacity classes are listed in Table 9.4. The interest levels are classified into high ($[0.75, 1]$), medium ($[0.25, 0, 75[$), and low ($[0, 0.25[$). The respective results for the varied virtual world velocity experiments are shown in Figures C.5, C.6 C.7, and C.8.

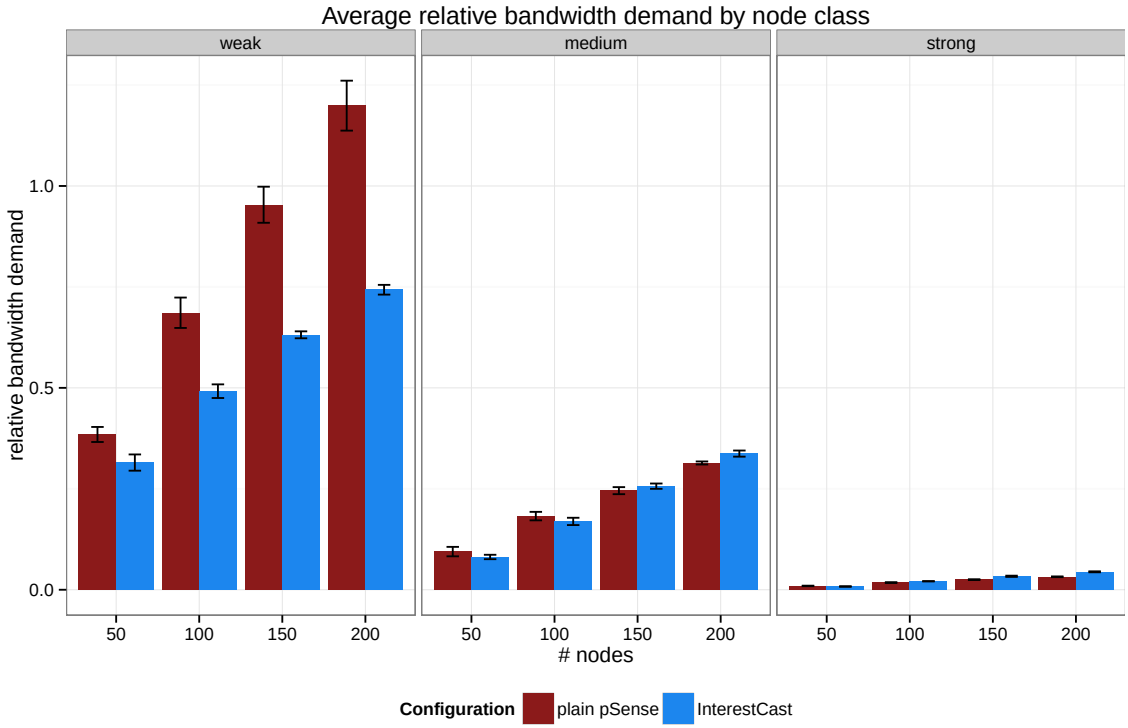


Figure C.1.: Average relative bandwidth demand by node capacity class depending on virtual world density. Error bars indicate the standard deviation of the measurements for five experiment runs.

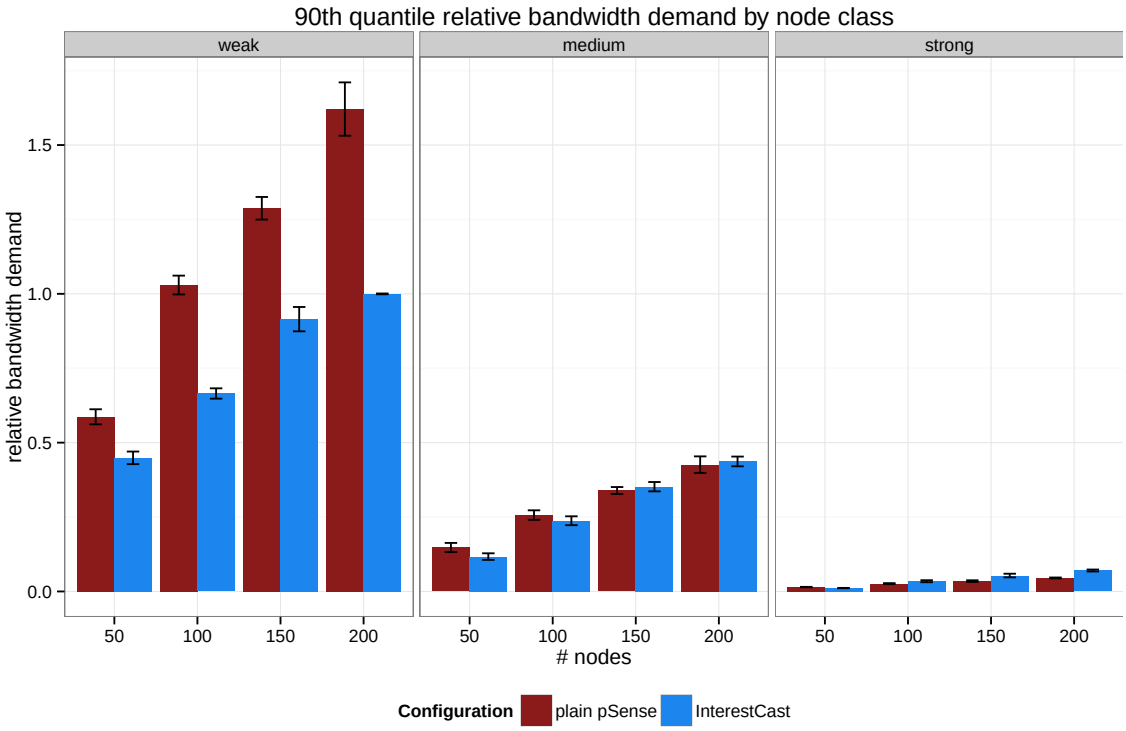


Figure C.2.: 90th quantile relative bandwidth demand by node capacity class depending on virtual world density. Error bars indicate the standard deviation of the measurements for five experiment runs.

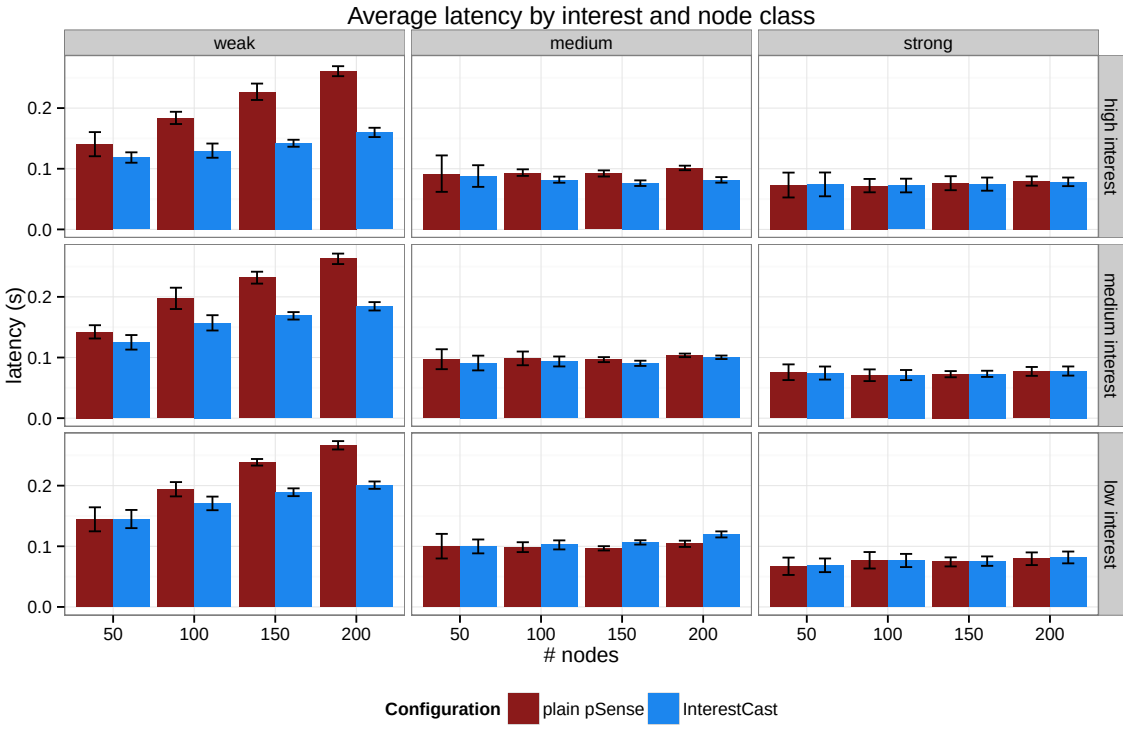


Figure C.3.: Average latency by node capacity and interest level class depending on virtual world density. Error bars indicate the standard deviation of the measurements for five experiment runs.

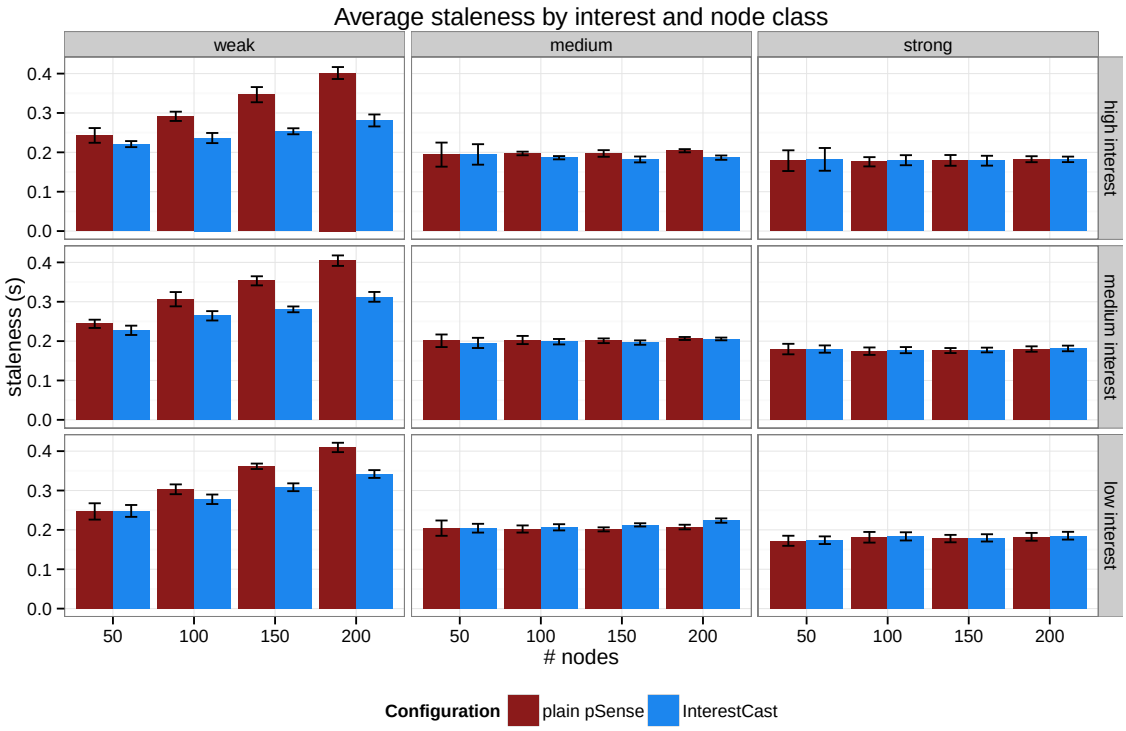


Figure C.4.: Average staleness by node capacity and interest level class depending on virtual world density. Error bars indicate the standard deviation of the measurements for five experiment runs.

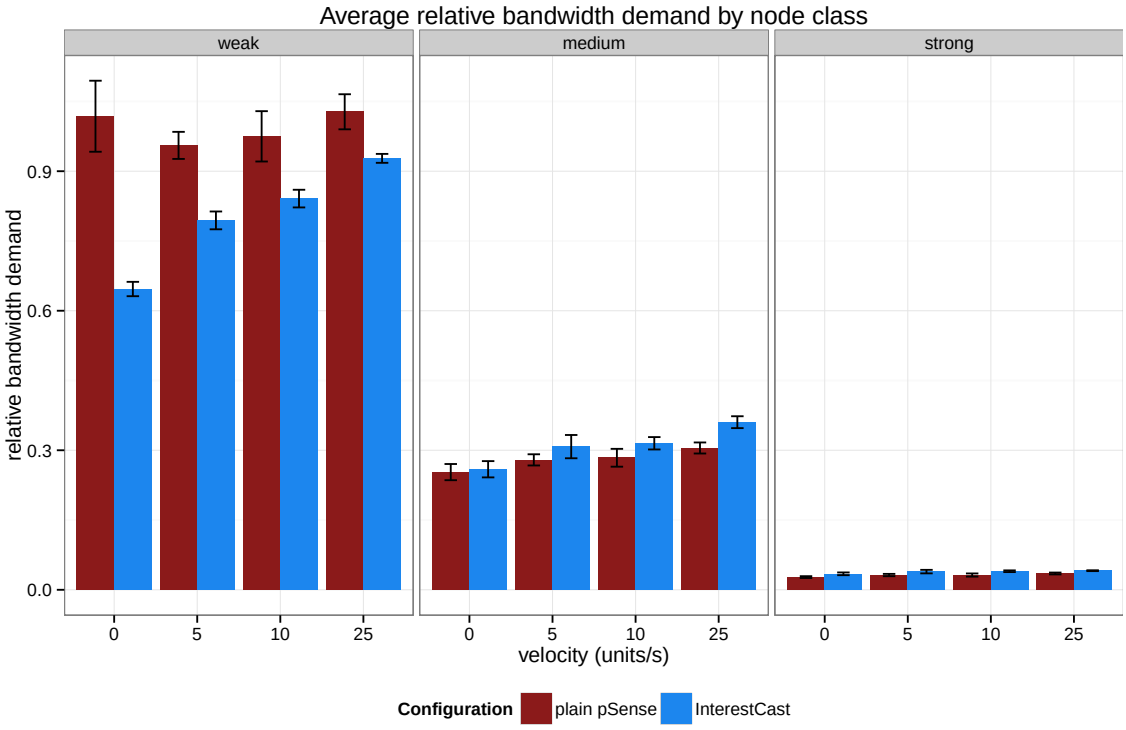


Figure C.5.: Average relative bandwidth demand by node capacity class depending on virtual world velocity. Error bars indicate the standard deviation of the measurements for five experiment runs.

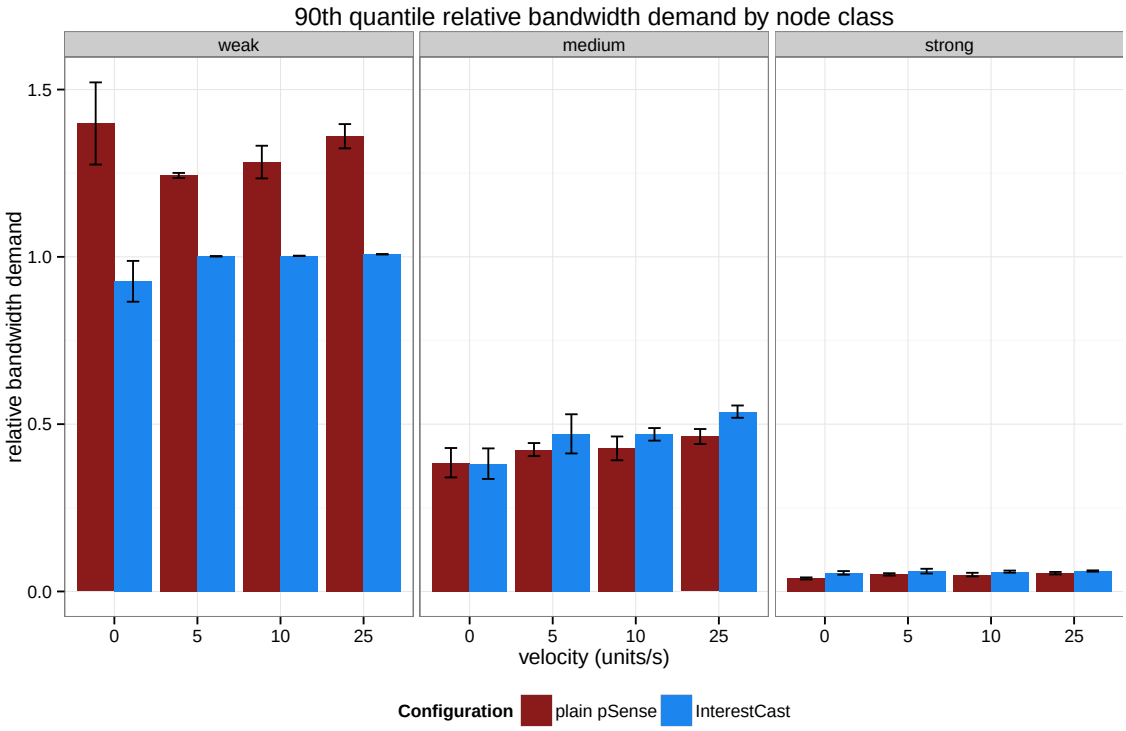


Figure C.6.: 90th quantile relative bandwidth demand by node capacity class depending on virtual world velocity. Error bars indicate the standard deviation of the measurements for five experiment runs.

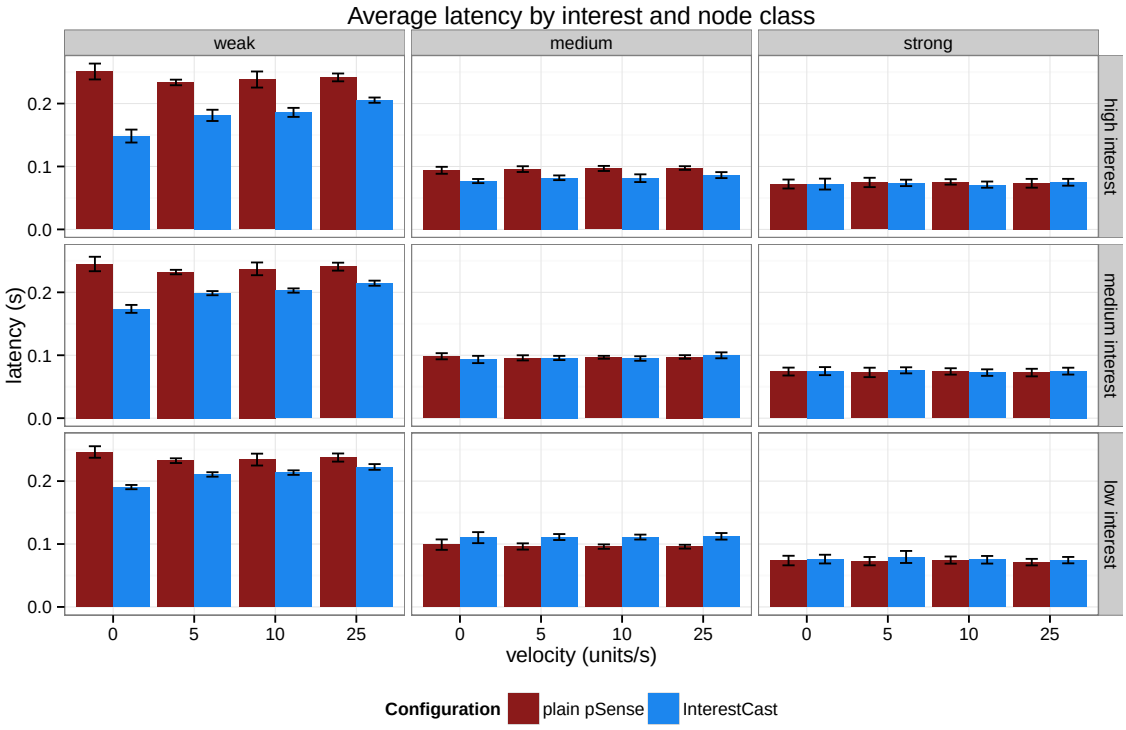


Figure C.7.: Average latency by node capacity and interest level class depending on virtual world velocity. Error bars indicate the standard deviation of the measurements for five experiment runs.

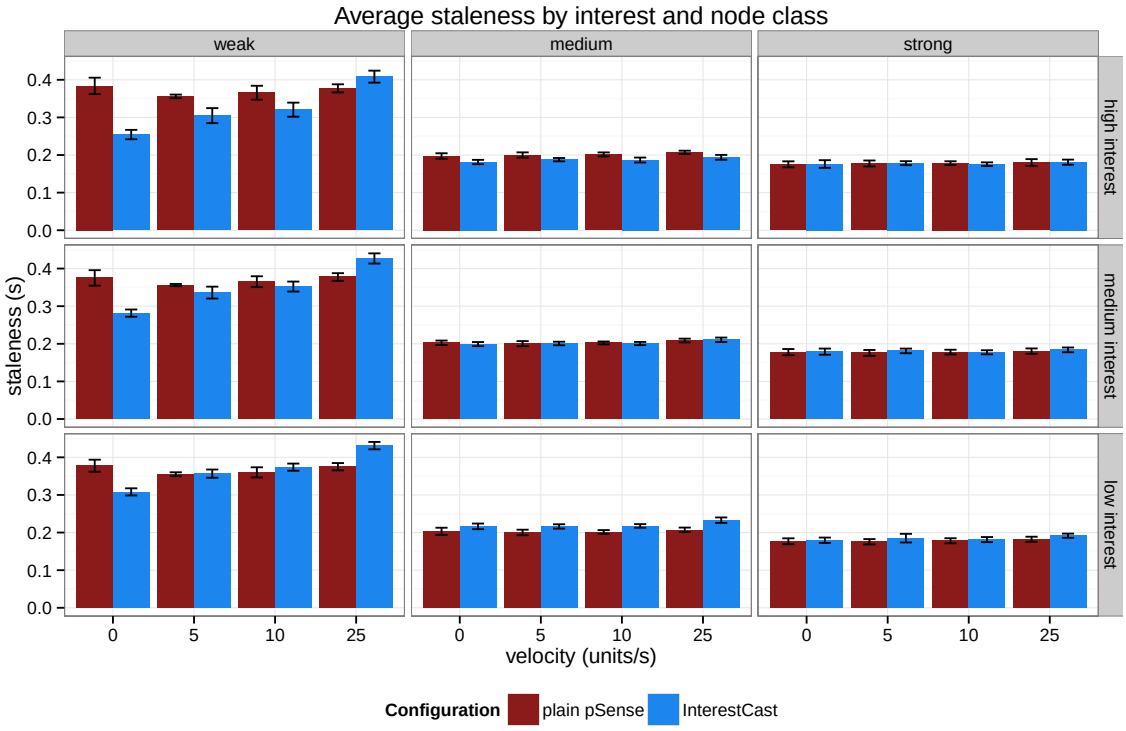


Figure C.8.: Average staleness by node capacity and interest level class depending on virtual world velocity. Error bars indicate the standard deviation of the measurements for five experiment runs.

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[2] gemäß §20 Abs. 3 der Promotionsordnung der TU Darmstadt